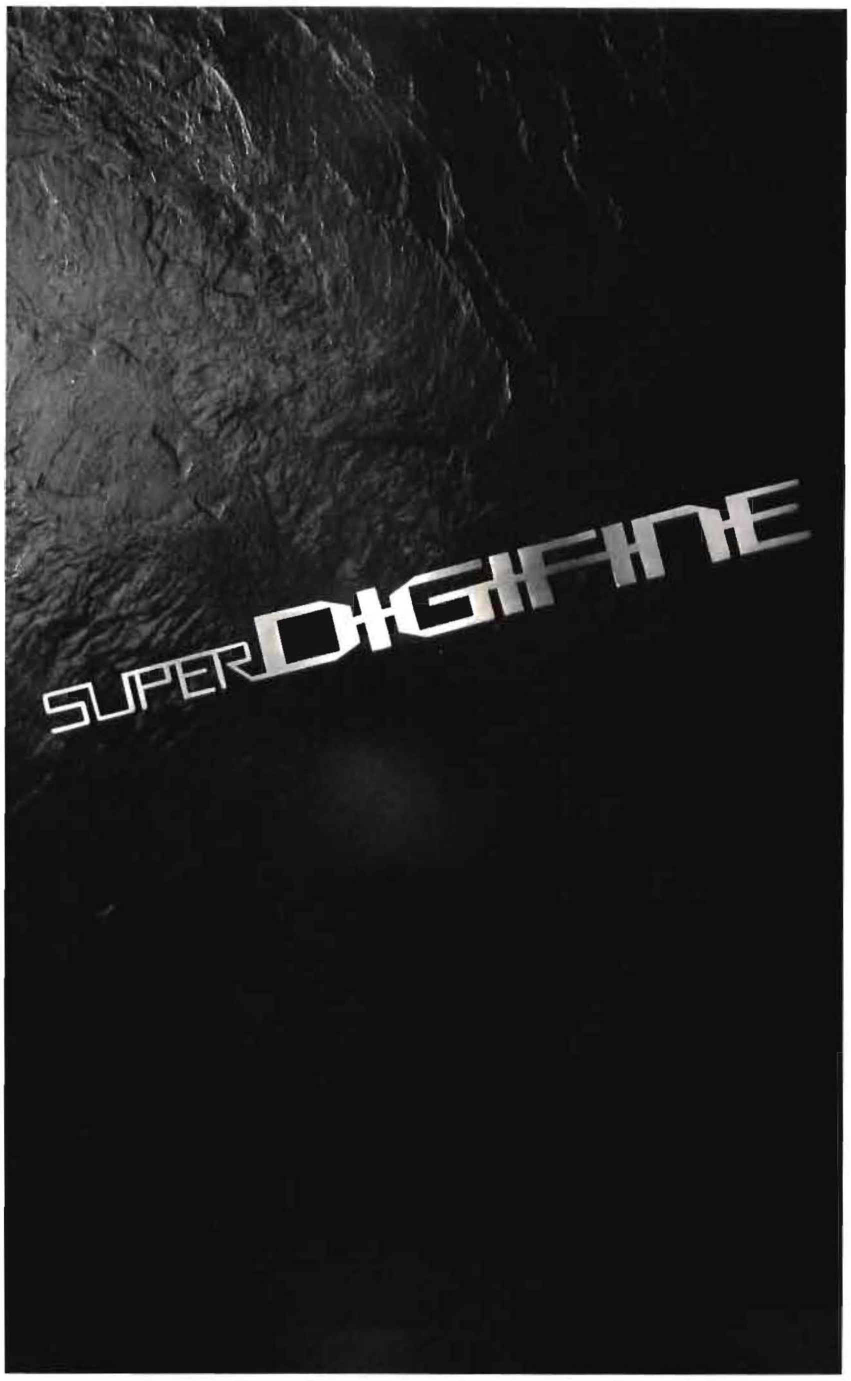
# JVC SUPER DIGIFINE HI-FI COMPONENTS



# SUPER DIGIFINE

*The JVC SUPER DIGIFINE series of hi-fi components are standard-setters that define the shape of things to come in audio. The series includes not only high-performance digital equipment but also components designed to take advantage of the dynamics of digital sound. What's more, as befit our top-of-the-line hi-fi components, each is finished in classy titanium gray and features exceptionally clean design.*

*Enjoy digital reality the JVC way—with SUPER DIGIFINE hi-fi components.*


JVC
Programmable Remote Control

# XL-Z1010TN
# COMPACT DISC PLAYER

## *JVC's advancing D/A conversion technologies bring you more delicacy and subtlety*

The heart of a digital processing system, like that of a CD player, is the D/A (Digital-to-Analog) converter. It's no exaggeration to say that it's the D/A converter that can make or break a CD player, determining the precision, accuracy and sound quality. That's why JVC offers two solutions to the problems that jeopardize the accuracy and precision of D/A conversion. Our K2 Interface eliminates "ripple" and "jitter" that can harm sound quality by transmitting digital codes, and codes alone, in their purest possible form. The Quadruple Full-Time Linear 18-Bit Combination D/A Converter combines high resolution with the ability to precisely reproduce subtle nuances.

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Product

**Mixing console at JVC Musical Industries Aoyama (Tokyo) Recording Studios**

## *The K2 Interface: a revolutionary "pure" signal transfer system*

### Background

Once we assumed that so long as binary codes, or bits (1s and 0s), were correctly converted, stored, transmitted and retrieved, digital signals would not change, even in the slightest.

But studio engineers and artists involved in digital recording had a different story to tell—they found that there were changes in sound quality; changes that were noticeable when these recordings were mastered, pressed into a disc and played back.

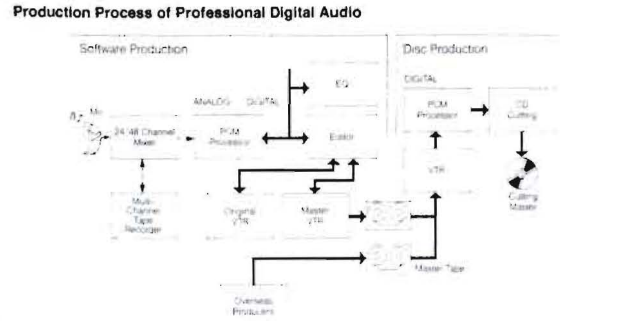
What our engineers have discovered is this: when a recording is dubbed from one tape to another, there is a change in sound quality, despite the fact that no analog processing is involved. Indeed, sound quality can change not only when tapes are dubbed but also when a different master tape is used, when a different master recorder is used, or when recordings are edited or equalized on a master console. This phenomenon occurs along the routes connecting various equipment whenever digital transmission is involved. So it's no wonder that the master tape and the discs produced from it sound different, or that the discs made from a master produced in one country sound different from the discs produced in another.

Audiophiles are also finding that using different digital connection cables between the CD player and an amplifier can cause differences in sound quality. And the same thing happens when a digital code is transmitted inside the CD player from the digital section to the analog. It also occurs inside amplifiers that come with built-in D/A (Digital-to-Analog) converters as the digital signal is sent from the digital to the analog section; that is, at the interface between the digital filter and D/A converter.

The quality of digital sound, therefore, *does* change during processing—but why? We discovered the answer through a collaboration of engineers in JVC's musical industries and a professional digital recording system developing staff in which they explored the intricate digital world of binary codes. The result of this research is the K2 Interface—a revolutionary signal transfer system for pure sound reproduction.

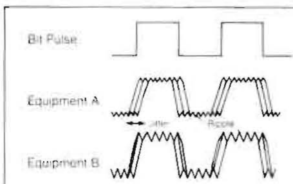
The implications of the K2 Interface are great, because it can be applied in digital audio and other digital equipment, whether the application is professional or consumer oriented. As a matter of fact, it can be used as a universal interface for digital equipment.

**Production Process of Professional Digital Audio**




### Ripple and jitter

As a digital signal is transmitted from the digital to the analog section, non-code components—components that are totally unrelated with the transmitted code—are generated inside the digital section and passed on to the analog section. It is these components, we have discovered, that cause the change in sound quality

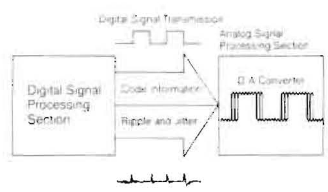



**Comparison of digital waveforms with different hardware**
Levels and shapes of ripple and jitter contained in a digital signal differ depending on equipment, which leads to changes in sound quality.

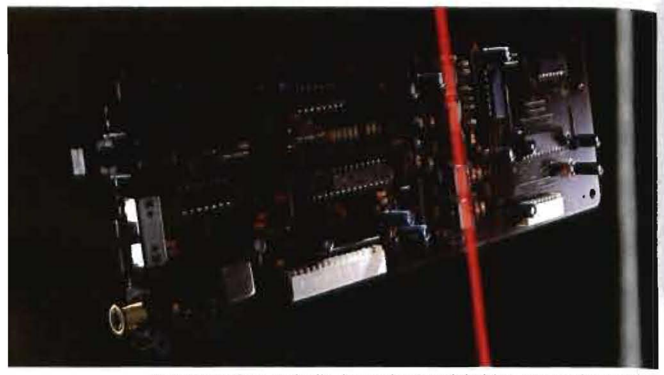
These unwanted non-code components are called ripple and jitter. Ripple is waveform distortion, whereas jitter is a shift in time. They are generated by load fluctuations in the power supply or stray capacitance and inductance of the circuits. As you can see in the figure above, non-code components are superimposed over bits for 1s and 0s and passed to the analog section for processing. Spectrally, a signal with ripple and jitter contains, in addition to the basic clock frequency, high-frequency harmonics that should not be there.

Moreover, digital circuits generate noise when their devices are in a transient state—switching in and switching out. This type of noise is generated because the change in voltage and current occurs in a finite time.

All this means that a digital signal contains both harmonic distortion and noise. When this signal is transmitted to the analog signal section, it causes various types of interference and adds intermodulation distortion in the analog signal. A digital signal may contain only a small amount of non-code components, but the distortion they cause contains harmonic components that are totally unrelated to the original analog signal. Therefore, in excessive cases, this type of distortion can even change the nature of digital music.



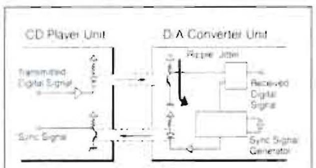

**Concept of conventional digital signal tranamission**
Non-code components (ripple and jitter) are passed to the digital signal processing section during transmission, affecting the transmitted signal and hence greatly changing the sound quality.



The K2 Interface optically decouples the digital from the analog sec ion

## Conventional optical transmission system

A conventional optical transmission system uses a waveshaping technique to remove ripple and jitter. This system is supposed to cut electric interference between the digital and analog sections, eliminate ripple and jitter and provide "clean" signal transmission, but in-use tests have shown that it doesn't. True, the transmitted signal looks clean on the scope, but it actually contains ripple and jitter—spurious harmonics that change the waveform. Once inside the analog section, these harmonics interfere with the grounding system and mix into the analog output, ultimately affecting sound quality.




**Conventional optical transmission system**
In a conventional optical transmission system, the digital signal is transmitted as waveforms, not as a code, therefore it cannot eliminate "non-code" components (ripple and jitter)

## The K2 Interface: the concept

At JVC, we tackled the problem of ripple and jitter from a completely new angle. The idea was to devise a design whereby codes, rather than waveforms, would be transmitted, and a totally new code would be generated by the receiver. In this way, non-code components can be eliminated from the signal transmitted to the analog section.

In the K2 Interface, the digital and analog signal processing sections are completely separated so that spurious non-code components cannot be transmitted from one section to the other. This system has a "transmitter" and a "receiver." In this design, the receiver looks at the transmitter at regular intervals for a brief enough period to know whether the code being sent is a 1 or a 0, and then generates the appropriate code on its own. That's the idea behind the K2 Interface.

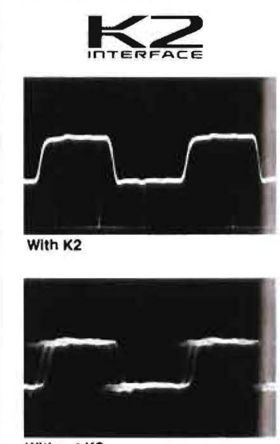
With the K2 Interface, coded digital signals are optically transmitted through a photocoupler from the digital processing section to the analog processing section. A code detection switch is connected to the ground of the photodetector, and is normally open. Therefore, the digital and analog sections are completely separated, making it impossible for non-code components to enter the analog section. Digital codes are controlled by the sync signal generated by the master clock in the analog section and sent to the digital section through a second photocoupler.

Coded bits from the CD are each about 708.6 nanoseconds (one nanosecond is equal to one-billionth, or 0.000000001 of a second) in length, and are precisely controlled by the sync signal from the master clock. With the K2 Interface, a trigger signal is sent from the master clock to the code detection switch twice for each code—that is, every 354.3 nanoseconds—which closes the switch for about 20 nanoseconds. Current flows during this short period only when the code is 0.

Since the code is detected at every half-code length, the detected signal is not affected by jitter. And since the switch is closed for only the briefest of moments—just 20 nanoseconds—the signal is not affected by ripple, either. Therefore, the spurious non-code components are not transmitted with the signal. Finally the detected code is sent to a D-type flip-flop and a new code—a pulse 708.6 nanoseconds in length—is generated according to the timing of the master clock. The output from the flip-flop then goes to the digital filter, the D/A converter and eventually to the analog circuitry.

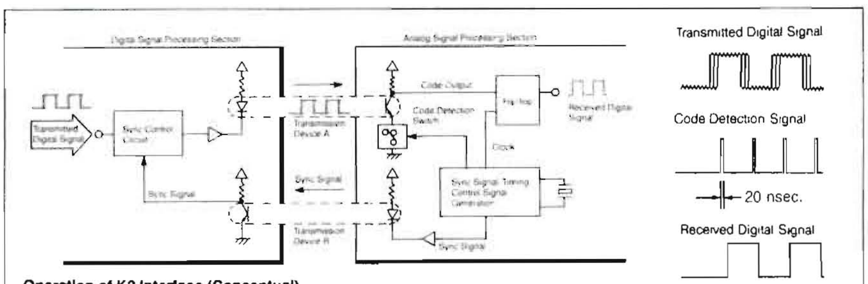
## Improved performance by K2 Interface

Here is proof of how the K2 Interface removes ripple and jitt r, reduces intermodulation distorti n and noise.

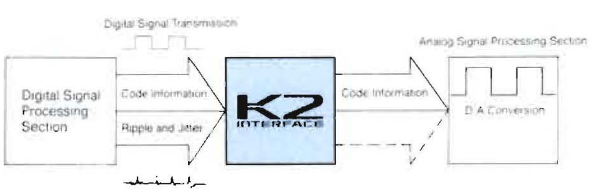



**Data Transmission Waveforms**

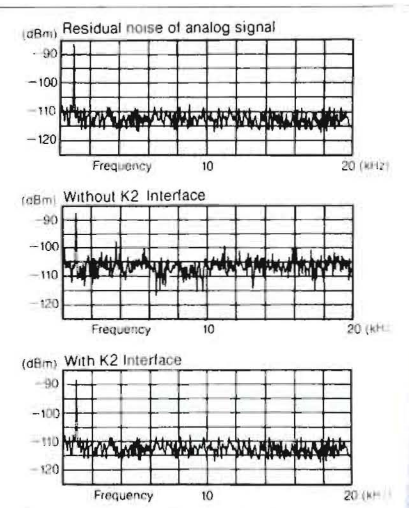
In the data above, it's appare t that the transmitted code by the K2 Interface is free of ripple and jit er.




**Operation of K2 Interface (Conceptual)**




**Concept of the K2 Interface digital transmission system**
With the K2 Interface, only code information is transmitted, therefore, unlike conventional transmission systems that use a waveform shaping technique, it does not change sound quality by adding ripple and jitter.




**Frequency spectra of analog output**

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The graphs on page 4 show the frequency spectra of residual noise contained in the analog output from a model with and without the K2 Interface. In this test, heavy loads are applied to the power supply for the transmitter in each model in order to induce non-code components—random noise—in the 1kHz transmission signal.

As you can see, when random noise is mixed with the digital signal in a model without the K2 Interface, it increases the residual noise level and changes its overall shape. This means that external disturbances applied to the digital section can affect the analog output and cause distortion so severe that it can alter the nature of the music. In the model featuring the K2 Interface, residual noise remains the same even if random noise is mixed with the digital signal. This is proof of the K2 Interface's ability to shut out external disturbances and prevent them from affecting the analog output.

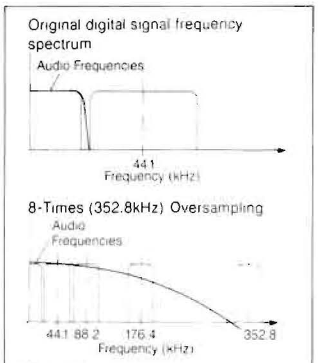
**Applications of the K2 Interface**
We have used the K2 Interface in our Compact Disc Player—the XL-Z1010TN. But because of its universal nature, it can also be applied in other digital equipment as well, like amps with built-in D/A converter (our AX-Z1010TN is one), DAT (Digital Audio Tape) decks, direct satellite broadcast reception or even professional equipment—indeed anywhere digital transmission is involved.

The digital sound processed by the K2 Interface is distinguished by better resolution, enhanced ambience and higher sense of realism it presents. We assure you thanks to the K2 Interface you will be enjoying the studio-quality digital sound at home very soon.

## *Quadruple Full-Time Linear 18-Bit Combination D/A Converter*

In CD playback, the ultimate quality much depends on the precision and resolution provided by a CD player's digital filter and D/A (Digital-to-Analog) converter. The Quadruple Full-Time Linear 18-Bit Combination D/A Converter is JVC's way of bringing about the best possible digital sound.

**Advantages of 8-times oversampling digital filtering**




**Digital filter response**

The digital code read by the pickup from a Compact Disc is sampled by frequency of 44.1kHz and processed for 16-bit requantization. This process generates not only audible frequencies but also spurious frequencies whose spectrum occupies the area just above 20kHz, the higher end of the audible range. These are frequencies that are multiples of the sampling frequency with a difference of −20kHz or +20kHz—24.1kHz or 64.1kHz, for instance. These spurious frequencies are usually eliminated by a low-pass filter with a sharp attenuation response and complex design to prevent the noise they can create. But too-sharp attenuation inevitably compromises sound quality.


In the XL-Z1010TN, we use a digital filter that re-samples the digital signal at a frequency 8 times the original—that is, 352.8kHz instead of 44.1kHz. This shifts unwanted frequencies to a much higher frequency range than normal. The advantage of using such a high sampling frequency is that the low-pass filter used in combination can have a gentle attenuation response. This means phase distortion is drastically reduced to provide you clear imaging of stereo sounds. Moreover, it eliminates requantization noise more effectively.

By using a higher re-sampling frequency, our digital filter reduces the sampling rate from about 22 microseconds to only a few. This improves the resolution in the time domain as well as conversion accuracy. Our digital filter internally processes 16-bit data with double precision accurate down to the 32nd bit. Then, before the processed data is output to the D/A converter, it's rounded to 18 bits for the subsequent D/A converter.

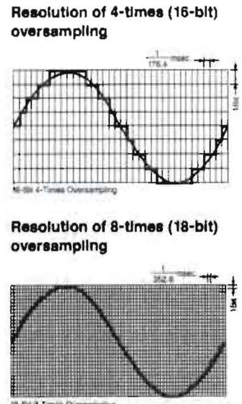
**JVC's Quadruple Full-Time Linear 18-Bit Combination D/A Converter**
The D/A converter ultimately reconstructs the oversampled digital code from the digital filter and converts it into an analog signal. For precise D/A conversion we developed the "Quadruple Full-Time Linear 18-Bit Combination D/A Converter" to operate in tandem with an 8-times oversampling filter.

Our advanced D/A converter features two D/A converter units for each channel—four in all. There is a 16-bit converter for most significant bits and a 2-bit converter for the two least significant bits. Since the least significant bits have greatest bearing on the sound quality at low levels, we use an elaborate discrete D/A converter system for these bits to ensure higher precision. Currents from these two converters are summed channel by channel, and the converters operate with 18 bits "full-time" whether the level is high or low.

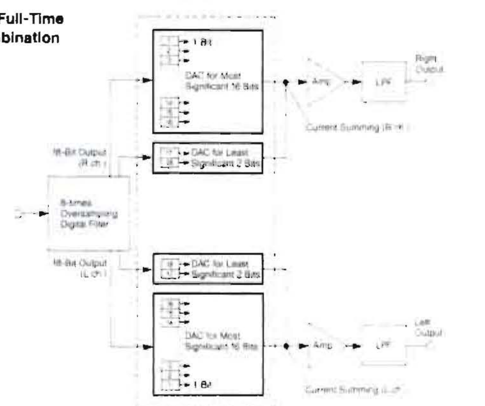
By combining an 8-times oversampling digital filter and 18-bit combination D/A converter, the digital signal processing circuit inside the XL-Z1010TN improves level resolution and time-domain resolution by 4 times and 2 times, respectively, over digital processing circuits using a 16-bit D/A converter with a 4-times oversampling digital filter.

All in all, our Quadruple Full-Time Linear 18-Bit Combination D/A Converter improves linearity, precision and resolution, allowing you to enjoy powerful and highly pure digital sound both at their most delicate or dynamic.

**Resolution of 4-times (16-bit) oversampling**
**Resolution of 8-times (18-bit) oversampling**




**JVC's Quadruple Full-Time Linear 18-Bit Combination D/A Converter**



## *Designs for quality digital sound*

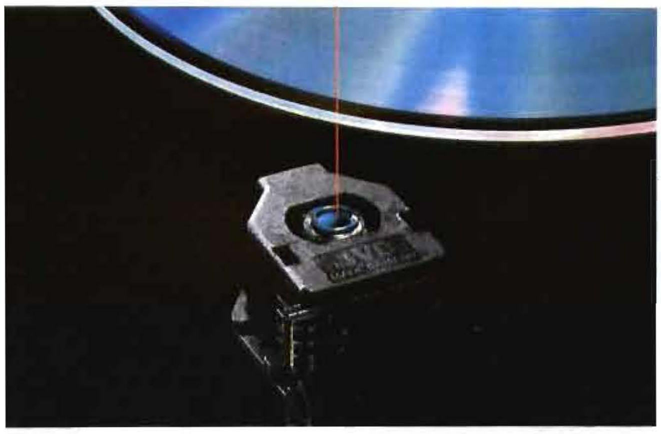
The digital signal processing section isn't the only part in the XL-Z1010TN we redesigned to improve sonic reproduction. We also took pains to improve other sections—including the analog section—in order to further refine overall performance. And the unit is mechanically constructed to prevent resonance, vibration and interference.

**New high-precision laser pickup design**

The XL-Z1010TN features a newly designed 3-beam laser pickup that combines high sensitivity, precision, stability and immunity to resonance and vibration. It's precise because the distance between laser beams has been shortened. And it's stable and resistant to resonance and vibration thanks to JVC's new suspended actuator. Our pickup is also compact and lightweight, improving tracking accuracy and reducing the noise caused by servo-controlling currents.

**Digital outputs—optical and coaxial**

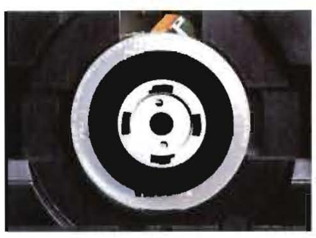
The XL-Z1010TN is equipped with two digital outputs—both optical and coaxial—so you can directly interface it with external digital components (e.g. the D/A converter in your amplifier or digital signal processing equipment). Using the optical output will electrically insulate the two connected components, shutting out digital noise that can compromise signal quality. What you get is purer, more musical digital sound.

**JVC high-precision 3-beam laser pickup**

**Disc stabilizing clamper**

Now that 3-inch (8cm) singles are available along with 5-inch (12cm) discs, the servo control in a CD player is given the new task of compensating for differences in weight, or moment of inertia, between two disc types by adjusting servo currents. This process can generate noise. Therefore, we've added a large stabilizing clamper to the disc rotating mechanism to equalize the moment of inertial for both types of discs. The clamper improves speed stability and keeps variations in servo currents and noise to a minimum.

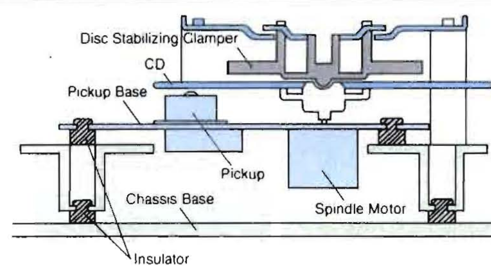


**New Y Servo for superior tracking ability**

Our "New Y Servo System" uses two special tracking beams—one leading and one trailing the main beam. The time difference between these signals is compensated for and the two signals are compared so as to cancel each other out. It is this sophisticated processing that enables the pickup to remain locked on the correct track—even when the disc is dirty or scratched. So the XL-Z1010TN's pickup does not "skip" tracks or repeat the same track over and over.

**Double-floating Independent Suspension System**

Subtle vibrations and resonance can degrade digital sound by affecting parts and devices inside a CD player. Therefore, in the

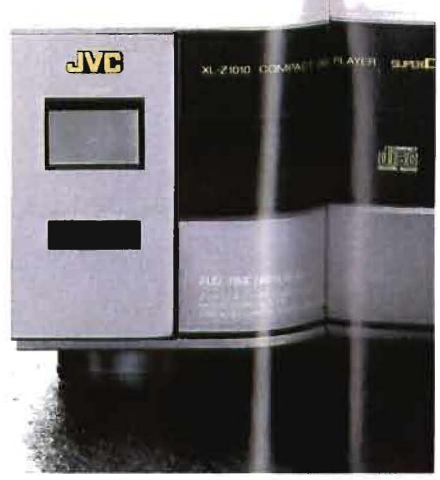



**Double-floating Independent Suspension System**

XL-Z1010TN, the pickup base is suspended from the mechanism base, and the mechanism base from the chassis—a design we call the "Double-floating Independent Suspension System." The result is that the pickup tracks a disc with highest accuracy despite shocks and sound pressure from speaker systems, thus ensuring clean reproduction unsullied by resonance and vibration.

**Separate digital/servo and analog circuits**

In most CD players, digital and servo-control circuits are laid out together with analog circuits. So the noise that servos generate during active operation or the clock pulse noise that microcomputers and digital circuitry create can easily affect the performance of analog circuits. In the XL-Z1010TN, however, we've separated the digital and servo-control circuits from the analog circuits, both physically and electrically. As a result, interference between digital and analog circuits is no longer a

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problem, and external noise no longer affects sound quality. Moreover, whenever desired, the fluorescent display can be turned off so the digital noise it generates will not compromise the music you hear.

**Separate digital/servo and analog circuits**

## *Designs for ease of use*

Another hallmark of the JVC XL-Z1010TN is its user-friendliness, for, after all, half the pleasure of dynamic digital audio is its convenience. And nowhere is that more evident than in our state-of-the-art CD player.

**3-way edit function: auto, programmed and multi-disc**
The 3-way edit function is a boon to those who are active in editing and recording CD tracks on tape. The "auto edit" mode lets you automatically fill a tape with as many selections from a disc as the tape can hold. Specify the length of time that one side of a tape can accommodate ("45" for a C90 tape, for instance), and the tracks are automatically recorded on tape—first side A and then B—in the same order as on the disc. The "programmed (manual) edit" mode lets you program tracks you want to transfer from disc to tape in any order. And as you do, you can assign the side on which each track will be recorded, A or B. And the "multi-disc edit" mode permits you to even program tracks from a number of discs for transfer to tape, assigning the side of the tape on which each track will be recorded. Using any of these modes, you can keep the unused blank space on a recorded tape to a minimum.

**Motor-driven volume control**
The XL-Z1010TN comes with a full-function remote control. As volume is adjusted from the remote, a motor is activated to drive the volume potentiometer in the main unit to adjust playback level. Its smooth, low-noise operation will not compromise sound quality as electronic volume controls often do. And there are two outputs on the rear panel—one with fixed and one with remote-variable level—for your operating convenience.

**Other features**
- JVC COMPU LINK Control System for interactive operation
- Multi-function display with 20-track program chart
- Headphone output and line ouput with volume control
- Random access programming of up to 32 tracks
- Intro-scan
- 4-way repeat (one track, all tracks, programmed tracks, A-B)
- Index play, skip and search
- Auto/manual search
- Remote control with volume control and numeric keypad


**Controls behind swing-down panel**

# RX-1010VTN
# RECEIVER/SYSTEM CONTROL CENTER

## *Intelligent design and a better user interface improve hi-fi operation*

The RX-1010VTN Receiver from the JVC SUPER DIGIFINE Series provides you with a look at the shape of a receiver in which computer control has been pushed to its logical limits. Most of its controls and operations are handled by microcomputers accessed via the handy remote control. Therefore, electronic switches are relegated to behind the swing-down door. And in the RX-1010VTN, our COMPU LINK Control System is represented in its most advanced and most convenient form, highlighting the new CSRP (COMPU LINK Source-Related Presetting).

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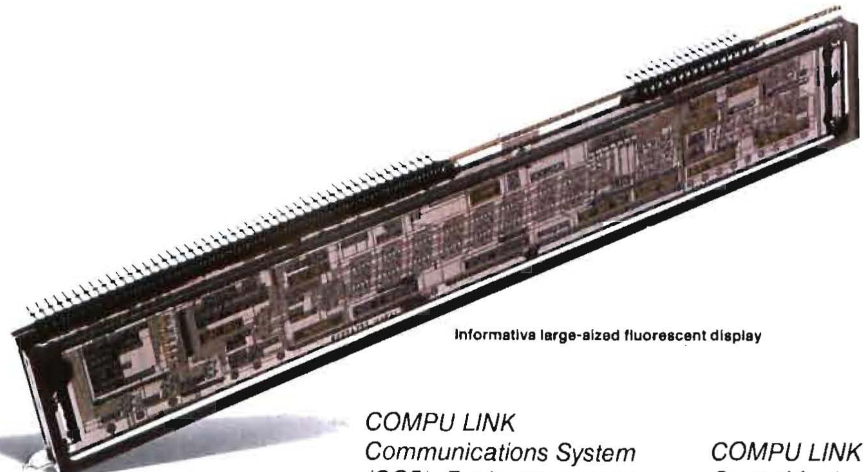


Informative large-sized fluorescent display

### *COMPU LINK Source-Related Presetting (CSRP): Acoustic memory program by program*

Up until now, you have had to manually adjust level, balance, equalization, and other parameters for the best sound each time you switch from one program source to another—touching up lows and highs as you change inputs from CD to FM, for example. But thanks to JVC's new COMPU LINK Source-Related Presetting (CSRP), you can program settings for the following parameters program source by program source or even for each of 40 FM/AM stations.


**Volume:** You can preset the volume program by program or station by station. This means the slight differences in sensitivities between components can be compensated for to ensure uniform output.


**Balance:** If you've finely adjusted channel balance for the best in your room, it can be committed to memory program by program.


**Equalization:** You can recall a preset or user-programmed equalization and store it in memory by program source or station by station. In this way, you can use a rather flat response for digital sound and a response with boosted ends for FM, for instance.


**Surround:** The surround mode and adjusted parameters may be preset for each program source or each preset station.

**Loudness:** The on/off position of the loudness switch can also be programmed source by source.

**DAP (Digital Acoustics Processor):** You can assign one of seven available ambience types (SYMPHONY HALL, RECITAL HALL, OPERA HOUSE, CHURCH, LIVE CLUB, STADIUM and MOVIE THEATER) to each program or each preset station.

Suppose that in setting these parameters, you've seen to it that when a video program is selected, volume is set at a moderate level; balance is set at center; an equalization for MOVIE is selected; Dolby Pro-Logic is turned on; and the "MOVIE THEATER" is set for digital acoustics processing. Then, at the mere touch of the VCR-1 button, you can be immediately transported into a world of incredible wrap-around sound. And this, of course, holds true for other inputs like CD, PHONO, TAPE and even each of 40 FM/AM preset stations.

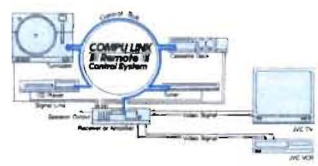

In addition, to improve ease of use, chosen parameters and digits are temporarily displayed on the fluorescent panel to help you check with the "CSRP TEST" or "CSRP DISPLAY" function and, if necessary, change the set parameters.

### *COMPU LINK Communications System (CCS): For better rapport between man and machine*

The CCS is another enhancement added to our COMPU LINK Control System, providing a better visual way to communicate between the user and the receiver. It graphically displays the mode of operation of a connected component in play, even for CD players and turntables. And it lets the user give a custom name, up to five characters long, to each FM or AM station or equalization that is preset and put into memory—"JAZZ-1" and "AM-3" for stations, "DISCO" and "POPS" for equalizations, for instance. Also shown are the chosen program source, volume level in digits, name of the Digital Acoustics Processor effect recalled from memory, and a host of other useful information. All you have to do to check the status of the receiver is just look at the display panel.

### *COMPU LINK A/V Remote Control for integrated audio/video*

With the JVC hi-fi components linked by COMPU LINK control bus line, you can control them from across the room using a single wireless remote control. So nearly every operation of the connected components—CD player, turntable, cassette deck, etc.—can be accessed right from where you sit. To add to convenience, even selected JVC TVs and VCRs can be directly controlled by the same remote unit, so that the user can tune a channel on the TV or the VCR, scan channels sequentially, and play or record on the VCR.



**Example of Fluorescent Display with CSRP Preset (VCR-1 as a source)**

(1) Source selected


(2) Volume, balance, loudness setting


(3) S E A graphic EQ setting


(4) Dolby surround setting


(5) DAP setting


(6) Normal setting

## *Programmable Remote Control—the ultimate in hands-off operation*

The clean looks of the RX-1010VTN owe much to the fact that many of the functions are electronic and therefore are consigned to the handy remote control. Moreover, this remote is a programmable one which can learn functions of other infrared remote controls.

Up until now, it has been necessary to keep handy all of the remote controls for individual audio and video components—a situation which tends to invite confusion. But our programmable remote control will replace nearly all the remotes for your audio and video components. It comes preprogrammed to control the receiver it comes with, of course, but also selected JVC components, like decks, CD players (even CD auto changers), turntables, VCRs and TVs.

And our programmable remote control also has the ability to learn control codes of most other infrared remote controls made by other manufacturers.* It stores the control codes for a total of 180 functions (should each use JVC's standard code length) in memory, allowing you to operate your audio and video components from a single unit.

In addition, our programmable remote comes with an LCD (Liquid Crystal Display) panel. As you select the desired program source, the display on the panel changes to show an appropriate menu screen showing only functions you need. Two modes are offered: the "standard" mode with 11 menus and the "programmable" mode with 14. For even more convenience, the symbols and numbers in each menu are not for display alone, they are actually switches that respond to your touch. So we've improved ease of use tremendously without adding more keys.

*Note: With our programmable remote controls, it may not be possible to program functions of infrared remote controls from other makers if they use different types of beams or if the codes they send contain too much information.

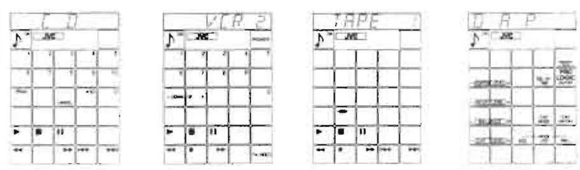

**Light button**
Provides back-lighting to improve readability in low-light situations.

**Indicators**
Indicators in this section simplify operation by telling you when a signal is sent or a function is learned, or when an error occurs during operation. Also featured here are a number of buttons for functions such as display screen pattern selection (SELECT 1), CSRP on/off (CSRP/CANCEL), component power on/off, and remote on/off.

**Programming the programmable remote control**
Our programmable remote control replaces most others, giving you the power to control nearly any remote-control components—regardless of make—in your integrated audio/video system. To program it, just flick one of its switches to LEARN, align our remote and an existing remote end-to-end, and press corresponding buttons on each. That's all!

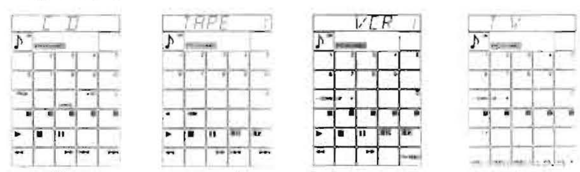
**Touch-panel LCD display**
The panel shows all the functions you've programmed for a particular input, but hides the functions you don't need. Displayed "keys" are actually touch-activated control buttons.

**JVC Standard Mode**



CD/CD Changer | VCR 2 | TAPE 1 | DAP/PRO-LOGIC

**"Programmable" Mode**



CD | TAPE 1 | VCR 1 | TV

Eleven and 14 menus are available for the standard and the "programmable" mode, respectively, including eight shown above.

**Display buttons**
Change the display without changing the selected program source.

**Source selection buttons**
When program sources are changed, the display on the panel also changes to show related functions.

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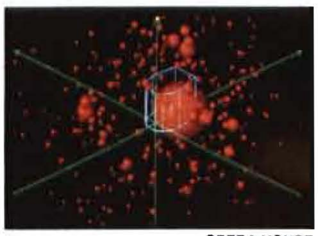
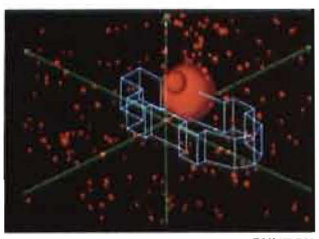
## *Digital Acoustics Processor*

The Digital Acoustics Processor, a technology based on our advanced digital engineering and our exclusive digital signal processing expertise, is built into the RX-1010VTN. It's a digital means of creating the ambience of a musical space—auditorium, concert hall, etc.—at home by applying a delay to sound. This system allows you to get the same exciting 3-dimensional sound you would at a live performance. Seven patterns of sound fields are resident in memory (SYMPHONY HALL, RECITAL HALL, OPERA HOUSE, CHURCH, LIVE CLUB, STADIUM and MOVIE THEATER), each available at a touch. You can also fine-adjust the ambience by controlling acoustic parameters such as ROOM SIZE, LIVENESS and WALL TYPE, so the created ambience sounds most realistic in your own listening room. JVC's precision computer-controlled acoustics measuring system called the "symmetrical 6-point sound field analysis method" and channel-by-channel acoustics processing guarantee you'll get a truly thrilling "you-are-there" feeling.



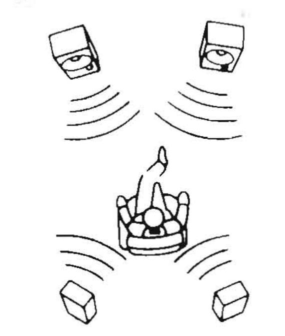
**OPERA HOUSE**



**CHURCH**

**Sound Field Analysis Patterns**

**Digital Acoustics Processor**



By just adding an extra pair of speaker systems, you can instantly upgrade to exciting live-ambience sound.

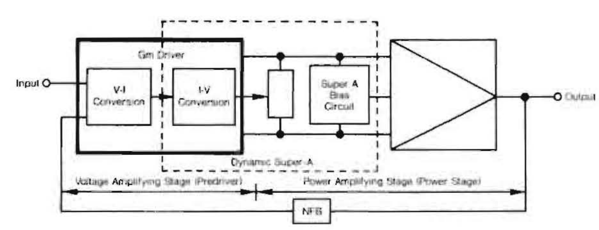
## *Dolby Pro-Logic Surround Sound*


The RX-1010VTN is equipped with the latest Dolby Surround sound decoder--the Pro-Logic Surround sound decoder. Using a new adaptive-matrix sound steering circuit, it enhances the sense of direction by boosting the output from the dominant channel and reducing the output in less-dominant channels. It also clearly localizes the dialog at the center channel, so conversation naturally comes from the screen. With the new Pro-Logic Dolby Surround, channel separation has been dramatically increased from mere 3dB to 25dB. Moreover, the RX-1010VTN's decoder uses digital delay circuit, therefore the circuitry will not degrade sound quality. All in all, the new depth and width you hear from video tapes and discs will add a new dimension to your video watching.

## *Dynamic Super-A for class-A sound*

Dynamic Super-A is a refinement of our original Super-A technology. With Super-A, a certain amount of bias, or idling current, is constantly applied to the power transistors to prevent them from switching off. The smooth, sinusoidal waveforms it provides are proof of the reduced harmonic distortion. Because Super-A does not generate switching distortion, it lets you enjoy low-distortion class-A sound. Dynamic Super-A further reduces distortion while improving the amp's overall response. Our Gm Driver has also been added to improve performance under actual in-use conditions.

**Output power: Stereo:** 120 watts per channel, min. RMS, both channels driven into 8 ohms, from 20Hz to 20kHz with no more than 0.007% total harmonic distortion;
**Surround;** 110 watts per channel, min. RMS, both channels driven into 8 ohms, from 20Hz to 20kHz with no more than 0.007% total harmonic distortion (front);
15 watts per channel, min. RMS, both channels driven into 8 ohms, at 1kHz with no more than 0.07% total harmonic distortion (rear).




**Block Diagram of Dynamic Super-A with Gm Driver**

## *High dynamic power*

Today, more dynamic power is demanded of amplifiers than ever, because of the wider dynamic range and better transient response of digital programs. So in the RX-1010VTN, we've improved the power supply so that it can instantly generate as much current as needed, especially when the impedance drops to 4 ohms or lower. Another way we've improved dynamic power is through the use of high-power, high-performance output transistors. The high dynamic power of 360W x 2 at 2 ohms that our receiver can provide is an indication of how it's ready to drive low impedances as presented by the combination of quality speakers and high-quality program sources.

## *Superb ease of use*

Over the years, the functions of our receivers have become more and more complex while their versatility has improved immensely. With the RX-1010VTN, we've simplified operation with the help of computers, electronic control and remote control—the reason why the front panel of the receiver is clean and uncluttered, despite its awesome control capabilities.

**Computer-controlled 7-band S.E.A. graphic equalizer**
The JVC S.E.A. graphic equalizer is a popular choice among audiophiles. With it you can change the sound to suit your own tastes, to overcome the acoustic deficiencies of your room, or to create a special sound for headphone or in-car listening. This is because a graphic equalizer allows you to adjust limited frequency ranges without affecting others. Besides, JVC S.E.A. graphic equalizers are known for accuracy, low noise and distortion and wide dynamic range.

We've built a 7-band graphic equalizer into the RX-1010VTN to give you the immense power of sound equalization. And to make it easier to use, we've computerized its operation. Here's what computerized equalization does for you.


**Examples of 7-band graphic equalizer indications (Frequency and control range)**


**Audio/video inputs and outputs on back (S-video terminals included)**

•**5 "namable" user-preset (manual) equalization:** You can create and store five custom equalizations, giving each the five-letter name of your choice (even your own name!).


•**5 programmable equalizations:** Five equalization patterns called HEAVY, CLEAR, SOFT, MOVIE and VOCAL are resident in memory.
•**Reverse equalization:** You can reverse an equalized response at a touch, say, to "compand" (compress/expand) a tape recording to reduce hiss noise.
•**Remote control:** With a remote in hand, you can equalize the sound and recall a preset equalization from where you sit—for the ultimate in equalizing ease.

**Computer-controlled tuner**
By combining a microcomputer with a PLL-quartz digital synthesizer tuner, the RX-1010VTN is superbly easy to use and accurate.
•**Preset memory:** You can preset a great total of 40 FM/AM stations for instant recall at the touch of a button.
•**Auto memory:** You can have the receiver preset 40 FM/AM stations all automatically for you.
•**Preset scan:** All 40 preset FM/AM stations in the memory can be automatically sampled for about 5 seconds each, helping you find the station you want.
•**Station name display:** You can assign a name of up to five alphanumeric characters to each preset station—"JAZZ" and "ROCK1," for instance—for easy identification.

And remember: The CSRP lets you customize acoustic parameters (equalizer settings, balance, volume, etc.) and keep them in memory, station by station.

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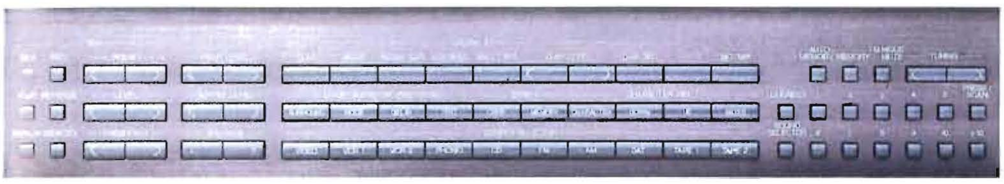
**Informative large-sized display**
A highly visible, large fluorescent display keeps you informed of the function you've chosen with the RX-1010VTN. The display is extremely easy to read, because all indicators are consolidated into logical groups.

**Audio/video integration**
The RX-1010VTN is ready to integrate your audio and video components into one easy-to-use home entertainment system—conveniently controlled from a single remote.
●**3 video connections:** Up to three video components (VIDEO, VCR-1 and VCR-2) may be connected and switched in and out. You can also dub from disc to tape or tape to tape. Moreover, there are S-video terminals in parallel that have separate pin contacts for chroma (C) and luminance (Y). By connecting TVs and VCRs (S-VHS VCRs) with matching terminals, you can enjoy a better picture and more accurate color.
●**Monitor output:** A monitor output lets you connect a monitor TV. There's an additional S-video terminal for a monitor with matching separate-chroma/luminance terminal.
●**Sound selector:** The selector lets you mate the audio of your choice with any video, a handy feature when you make your own video productions or "ambience" videos.

**Other features**
- High-gain phono equalizer for MM/MC cartridges
- Connections for 2 pairs of speaker systems
- Sleep timer



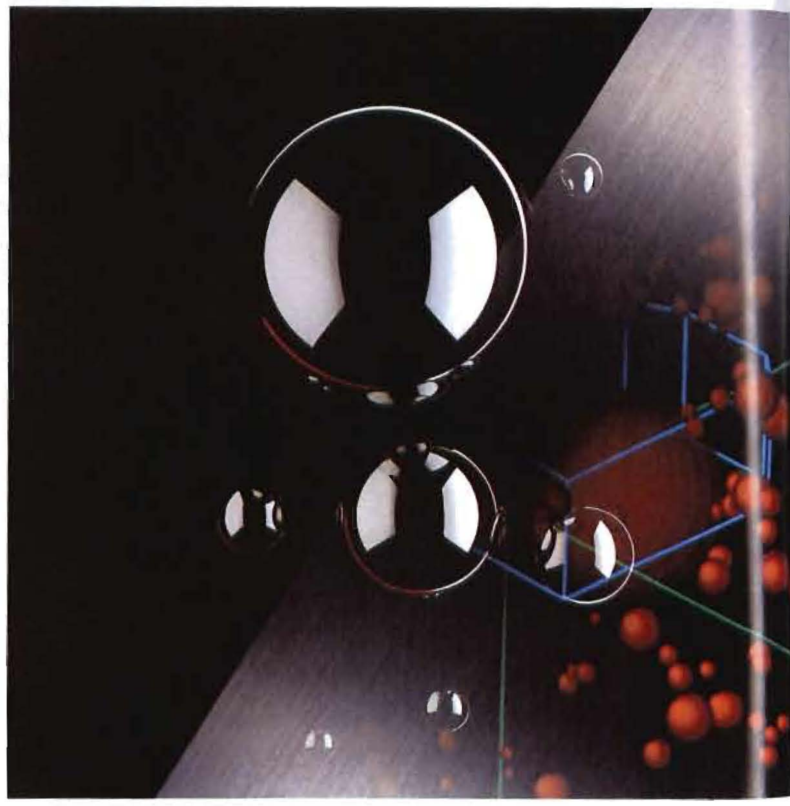
Controls behind swing-down panel

COMPU LINK Remote Control Component

# XP-A1010TN
# DIGITAL ACOUSTICS PROCESSOR

## *The digital way to bring the live musical experience home*

Recorded performances you hear on your stereo system sound fine, especially if they're digitally made, but the same piece of music played by the same artist sounds more exciting and more engrossing when you hear it live. But this truly realistic sense of "being there" is hard to reproduce on a conventional hi-fi system. The JVC XP-A1010TN Digital Acoustics Processor, however, is by no means conventional: it lets you enjoy the incredibly realistic ambience of a live concert right in your own home.



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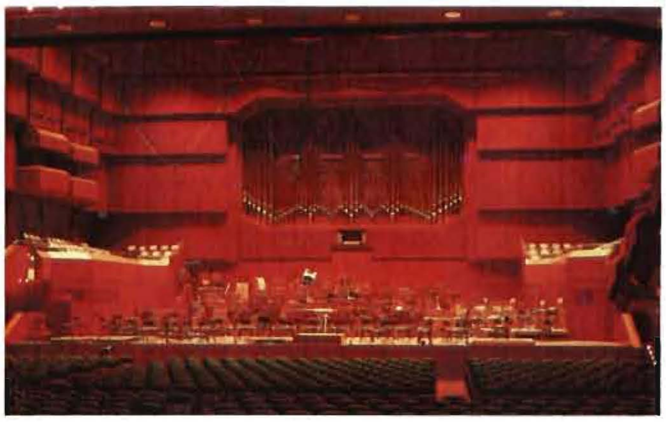


Frankfurt Alte Oper (Opera House)

Symmetrical 6-point sound field measurement microphone set

## What's "liveness"?

Suppose you're in a hall listening to a live musical event. When a note is played, the first thing you hear is the direct sound from the source. Direct sounds are then followed by "early reflections"—a group of sounds that are reflected by the walls, ceiling and stage with decreased volume. Finally what reaches your ears last are the "reverberations" which come from random directions over a relatively extended period. It's this pattern of sounds, reflections and reverberations that lets you know you're experiencing a live performance.

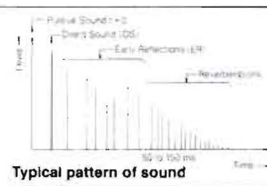
Each acoustic space (a hall or auditorium, for instance) has its own individual ambience, or the pattern of reflections and reverberations. When a space is large and acoustically "live," for example, intervals between individual early reflections are rather long, and reverberations decay slowly. The opposite is true for a small-sized room with "dead" acoustics.

## Digital acoustics simulation

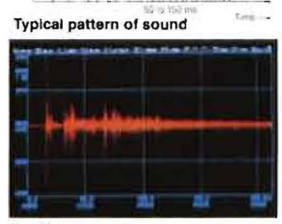
When a signal is turned into digital form, it can be delayed or otherwise processed without its quality being compromised or degraded. And computers have the amazing ability to perform complex analyses of a vast amount of data at very high speeds thanks to efficient VLSIs (Very Large Scale Integrated Circuits).

With this combination of digital technologies, we can now generate reflections of a concert hall with exceptional accuracy—by measuring precisely the ambience components in the sound heard in a hall, determining the direction and level of each, and then recreating this complex aural environment in the listening room, all electronically and digitally. In this way, it is possible to create an incredibly realistic ambience right in your own listening room.

But while digital technology has given us the awesome ability to turn a listening room into a concert hall, there are two important factors we must be aware of. One, the measured data on the acoustic characteristics of actual halls that will go into the memory of a digital acoustics processor must be accurate and comprehensive; otherwise, precise reproduction is impossible. And two, to synthesize the ambience of a hall accurately, two things must be considered—the ambience contained in the recorded music and the ambience of the listening room where the recorded music will be played back.



Typical pattern of sound

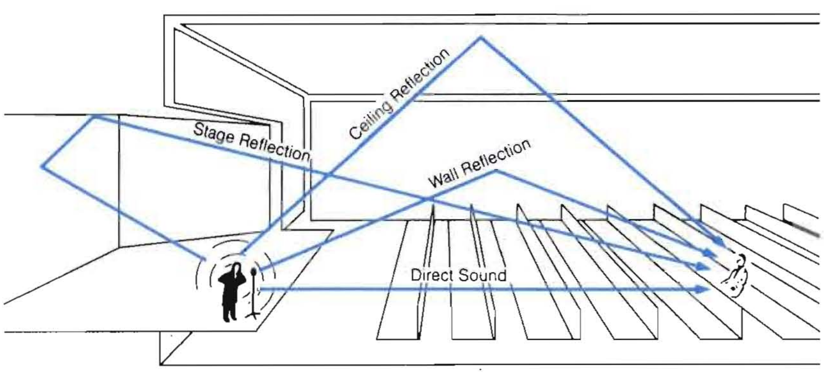


Impulse response

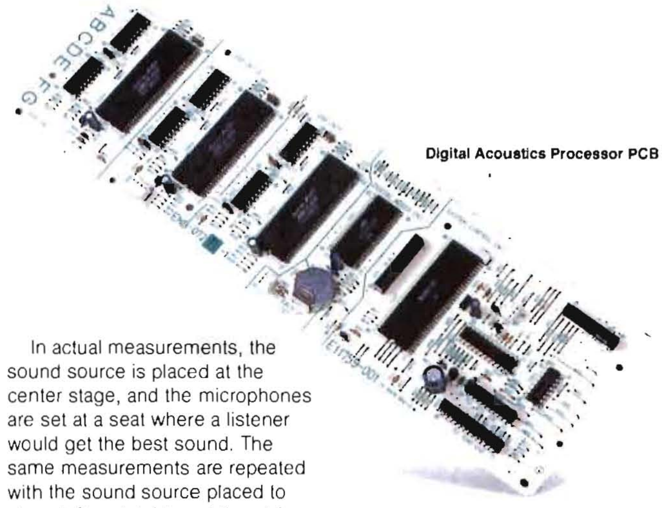
## JVC symmetrical 6-point sound field analysis method

To obtain accurate data on the ambiences of existing halls, we first developed a computerized system to measure them and quantify them—the "symmetrical 6-point sound field analysis method." With this system, a starter's pistol is used as the source to create the direct sound, since the noise it generates is pulsive. It's shot in a hall, and the sound and the reverberations it creates are picked up by a JVC-developed device using three pairs of precision-calibrated omnidirectional microphones set along X, Y and Z axes, equidistant from the reference point.

The signal picked up by the microphones are digitally recorded by JVC's Digital Audio Mastering System, and the components that comprise the picked-up sound are instantly shown on a display, with the resulting data stored on a floppy disk for later analysis




Acoustic response of a musical space

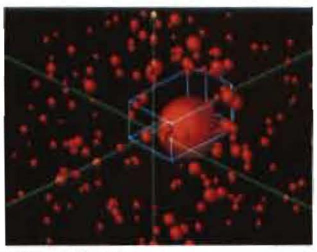


Digital Acoustics Processor PCB

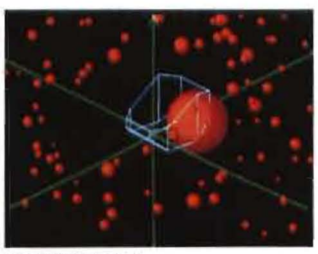
In actual measurements, the sound source is placed at the center stage, and the microphones are set at a seat where a listener would get the best sound. The same measurements are repeated with the sound source placed to stage left and right, and the pickup system is placed at the best seat in the house. Unlike conventional measuring systems, ours can measure the ambience of a small space, like that of a listening room. This has made it possible for us to compensate for the ambience of the listening room in the Digital Acoustics Processor.

To collect data that would go into our Digital Acoustics Processor, our engineers took the trouble of traveling all over Europe to measure the acoustics of famous halls there using our elaborate measuring system. We went to this trouble because we know that the simulation of a hall cannot be any more accurate than the data on which the simulation is based.

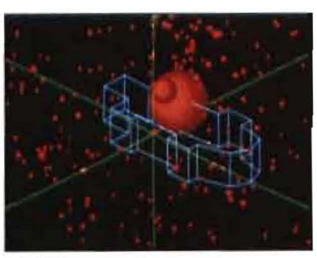
The data on the sound source and reverberations measured by the symmetrical 6-point sound field analysis method are displayed as 3-dimensional patterns for closer examination—each of the reverberations represented as a "virtual image source" along with its size and location. In this data, the center shape represents the floor of a hall. The largest circle is the direct sound, while others are "virtual image sources"; the farther away from the center, the longer delay they are given. Level is expressed by the size of a circle.



SYMPHONY HALL 1



SYMPHONY HALL 5



CHURCH

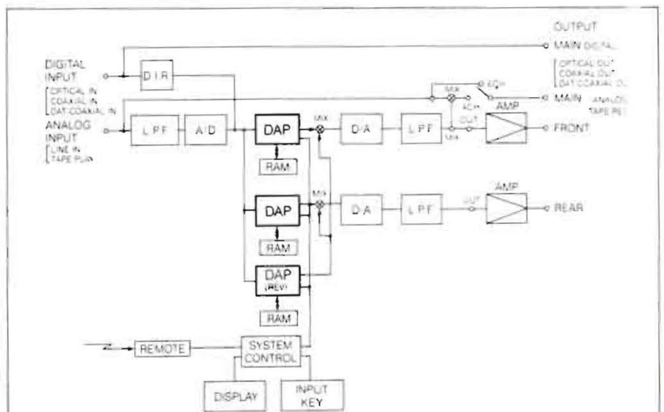
Sound Field Analysis Patterns

## *Features of the Digital Acoustics Processor*

In order to accurately synthesize the ambience of a hall at home, our Digital Acoustics Processor compensates for spurious ambiences that are generated as recorded music is played back in the listening room—the ambience contained in the recorded music and the ambience of the listening room itself. This is because the "virtual image sources" created by our digital processor will add their own reflections by bouncing off the walls, floor and ceiling in the listening room. So these must be first cancelled out. The same is true of the reverberations contained in the recorded music. If they are not fully compensated for, however, excessive reflections and reverberations can be added, totally ruining the sense of realism.

Also our digital processor gives the user the choice of "solo" or "spread" according to the size of the musical source, to recreate the ambience of a hall more precisely. The "LISTENING ROOM SIZE," "LISTENING ROOM REVERB," "SOURCE REVERB" and "SOURCE POINT/SPREAD" controls allow the user to adjust these parameters to get a desired ambience in any listening room from any program source.

Further, the JVC Digital Acoustics Processor contains 20 different measured ambiences in memory. This lets you instantly recreate ambiences of some typical musical spaces. And, to let you alter a pattern you've chosen more to your liking, it has controls to adjust six parameters for a desired ambience—ROOM SIZE, LIVENESS, LOW PASS FILTER, REVERB LEVEL, HIGH FREQUENCY REVERB, and OFFSET DELAY.




Block diagram of Digital Acoustics Processor

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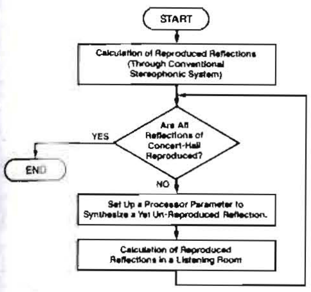
## *The process of simulating hall ambience*

In our Digital Acoustics Processor, a digital-processing VLSI (Very Large Scale Integrated Circuit) has the function of synthesizing the "virtual image sources" by digital processing for playback in a user's listening room. Virtual image sources in a "target" hall are synthesized in the following way.

**Flowchart of reverberation synthesis**




First calculations are made for reflection components generated in the listening room when the main speaker reproduces a direct sound, and a comparison is made between the resulting data and the ambience data for the target hall. If a component is contained in both data, then it is removed from the ambience of the hall.

Then, the largest reflection component in the hall ambience is selected. And calculation is made on a reflection generated when this reflection component is synthesized by two ambience speakers in the listening room. Finally, the resulting data is compared with the data for the target hall—if it is duplicated, it will be removed.

The process will be repeated for the 2nd largest reflection component in the hall, the 3rd largest, and so on, within the limits of the hardware. Into this loop of eliminating spurious ambience components is incorporated the environment set by parameters for each of the listening room, the sound source and the target hall (LISTENING ROOM SIZE, LIVENESS, etc.).

Moreover, in our Digital Acoustics Processor, digital processing of ambience is performed in stereo rather than in mono as in the past. By making better use of the ambience components contained in a source—including out-of-phase left/right information—it's possible to create the ambience of the target hall more accurately and provide a fuller musical experience.

**20 Programmed Sound Field Patterns**

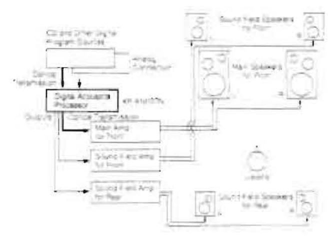
| No. | Program Name | Type | No. | Program Name | Type |
|---|---|---|---|---|---|
| 1 | SYMPHONY HALL 1 | SHOEBOX TYPE | 11 | LIVE CLUB 1 | JAZZ CLUB |
| 2 | SYMPHONY HALL 2 | SHOEBOX TYPE | 12 | LIVE CLUB 2 | DISCOTHEQUE |
| 3 | SYMPHONY HALL 3 | SHOEBOX TYPE | 13 | PAVILION | LIVE CONCERT |
| 4 | SYMPHONY HALL 4 | VINEYARD TYPE | 14 | GYMNASIUM | HARD-FLOORED HALL |
| 5 | SYMPHONY HALL 5 | VINEYARD TYPE | 15 | STADIUM | OUTDOOR LIVE CONCERT |
| 6 | SYMPHONY HALL 6 | VINEYARD TYPE | 16 | MOVIE THEATER 1 | SMALL SPACE |
| 7 | RECITAL HALL | SMALL MUSICAL SPACE | 17 | MOVIE THEATER 2 | MEDIUM-SIZED SPACE |
| 8 | OPERA HOUSE | WITH TIERED SEATING | 18 | MOVIE THEATER 3 | LARGE SPACE |
| 9 | CATHEDRAL | GOTHIC STYLE | 19 | MOVIE THEATER 4 | EXTRA LARGE SPACE |
| 10 | CHURCH | HIGH-CEILINGED SPACE | 20 | MOVIE THEATER 5 | STANDARD |

**Variable Acoustic Parameters**

| No. | Parameter | Adjustable Range | Step | Initial Value | No. | Parameter | Adjustable Range | Step | Initial Value |
|---|---|---|---|---|---|---|---|---|---|
| 1 | ROOM SIZE | 0.5—2 | 0.1 | 1 | 7 | REAR DELAY | 15—30ms | 1ms | 20ms |
| 2 | LIVENESS | 0.5—2 | 0.1 | 1 | 8 | SPREAD/POINT | SPREAD/POINT | | SPREAD |
| 3 | LOW-PASS FILTER | 1—16kHz, THRU | 1kHz | | 9 | LISTENING ROOM REVERB | 0.2—0.6ms | 0.1ms | 0.4ms |
| 4 | REVERB LEVEL | 0—2 | 0.1 | 1 | 10 | LISTENING ROOM SIZE | 10m² or less, 10—16m², 16m² or more | | 10—16m² |
| 5 | HIGH-FREQ REVERB | 0.1—1 | 0.1 | | | | | | |
| 6 | OFFSET DELAY | 0—200ms | 1ms | 0 | 11 | SOURCE REVERB | 0—5 sec | 0.1 sec | 0 sec |

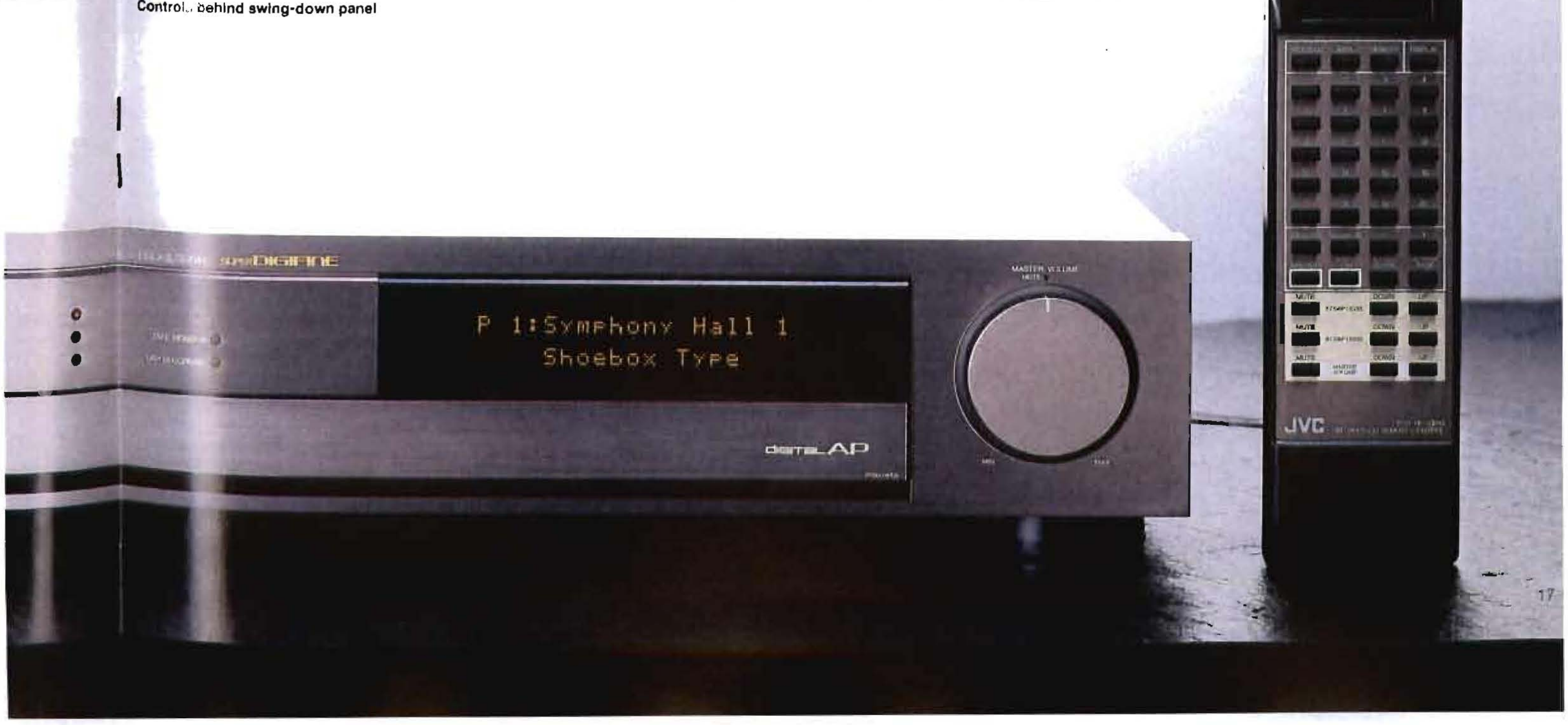
### Other features

- •Direct digital inputs and outputs: optical and coaxial
- •4/6-channel system configuration selectable
- •6-ganged motor-driven remote-control volume control
- •Programmable 5X7-dot 40-character fluorescent display



**System connection diagram**
The JVC Digital Acoustics Processor provides the most realistic ambience reproduction when a third pair of speakers are employed to add height

Controls behind swing-down panel



# AX-Z1010TN DIGITAL PURE-A INTEGRATED AMPLIFIER WITH K2 INTERFACE

## *Purity, dynamics and an enhanced sense of power*

What are the two most critical criteria demanded of an amplifier in this age of digital audio? High power and high purity. That's because it's the combination of these two factors that lets you enjoy digital sound to its fullest. Until now, getting both has been next to impossible. But now JVC brings you both high purity and high power with the help of sophisticated digital engineering and our Digital Pure-A II circuitry, both featured in the AX-Z1010TN integrated amplifier from the SUPER DIGIFINE series.

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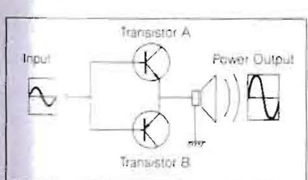
**Inside view of the AX-Z1010TN**

## *Digital Pure-A II*

### High purity vs. high power

The task of reproducing music with dynamic power and delicate purity is a daunting one for an amplifier. Of course, today's amplifiers cannot give you the full impact of live music, but we're getting closer. There is one way to ensure high purity in a hi-fi system--by using a class-A amplifier. The reason is that this design does not suffer from "switching distortion." The one drawback of class-A amplifiers, however, is that they are inefficient, requiring a massive power supply and wasting most of it in the form of heat. So up until now, class-A operation and high power could not exist in the same amplifier.

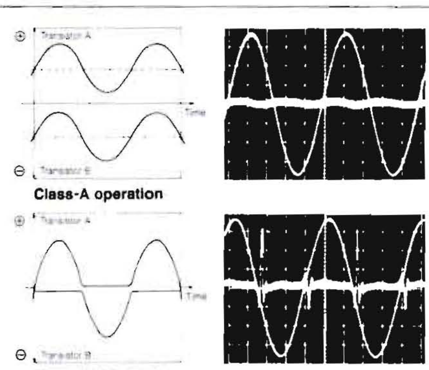
### Push-pull configuration




Most amplifiers use an output configuration called "push-pull" in order to improve linearity and efficiency of power transistors and also to achieve high power. In the push-pull design shown above, the transistor on top and the one below individually amplify the positive and negative side of an input signal to improve efficiency. The signals amplified by the top and bottom power transistors are combined in the output, and, driven by the power supply, are turned into power to move the speaker.

The power supply plays a critical role in any amplifier, especially if it's designed for better digital reproduction. The supply should feed enough power to the power transistors so that the transistors operate with high efficiency and reproduce both dynamic and delicate sounds with power to spare. In this respect, remember power (W) equals current (I) times voltage (E). This means, if the current is given, you'll have higher power by increasing voltage, and vice versa. From this it's apparent that a reduction in power will result if there's loss in either current or voltage.

**Class-A vs. class-B operation**
With class-A, combined output from the paired transistors looks very close to that of the input. With class-B, this is not so, leading to crossover and switching distortion.




**Class-A operation**

**Class-B operation**

**Operating Current Waveform** **Output Waveform**

### Class-A vs. class-B operation

An amplifier can be either class-A or class-B, depending on how the transistors in the push-pull arrangement amplify the applied signal. With class-B amplifiers, each of the paired transistors amplifies an input while it is conductive, but is switched off as it tries to amplify the side it's not responsible for. Because little bias is applied, this results in switching and crossover distortion, which impairs purity. Besides, the waveform of varying current is not as smooth as a sine wave. This means the output contains the kind of harmonic distortion that can be irritating to our ears. Yet the advantage of class-B amps is that since little bias is applied, excessive heat is not generated, so the amps do not need a bulky and costly heat-sinking system. For the same reason, since they are highly efficient, they can easily deliver a high power.

In class-A amps, a sufficiently high bias is constantly applied to the power transistors in order to accommodate higher dynamic levels. This ensures that the current waveform is much smoother and that the combined output looks closer to that of the input: this input/output symmetry is maintained at any level so that pure reproduction is achieved with extremely low amounts of harmful harmonic distortion. But since a high level of bias is applied to the transistors, excessive power is generated and much is dissipated as waste heat. This means a high-power class-A amp requires a prohibitively costly heat sinking system and power supply.

So all in all, what's the most ideal amplifier design? It's the combination of the two classes of amplifiers without disadvantages of each—one that combines the low distortion and smooth response of class-A with high power and efficiency of class-B.

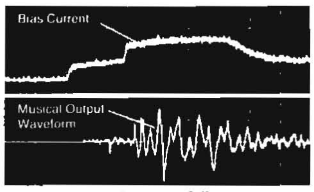
### JVC Digital Pure-A II

The JVC Digital Pure-A II operates just like the most ideal amplifier, offering the combined advantages of class-A and class-B. It's an amplifier that varies the bias according to dynamics of the input in order to combine high purity and high power: it supplies low bias when the input level is low, intermediate bias when the level is moderate, and higher bias when the level is high. In other words, Digital Pure-A II is an ideal amplifier that combines high power, high efficiency and pure sound. The innovative thing about this design is that it takes advantage of the unique nature of the digital signal: when in digital form, a signal can be delayed without degrading its quality.

Digital Pure-A II operates in the following way:

Digital signals fed directly from digital equipment (a CD player, for example) are split into two signals: the main and the "prediction" signals. The main signal is sent to a time base processor where it's stored in memory for a very brief 10 milliseconds before being passed on to the D/A (Digital-to-Analog) converter. The prediction signal is sent to a prediction signal processor and on to the high-speed optical bias controller. In the prediction signal processor, the level of the upcoming main signal is measured, and a prediction signal is generated. In the optical bias controller a control signal is generated by comparing and

analyzing the prediction signal and the D/A-converted main signal from the "output level detector," where the absolute peak level of the power transistors is detected. Finally, the control signal is sent to the bias circuit to determine which of three bias steps is to be applied to the power transistors.

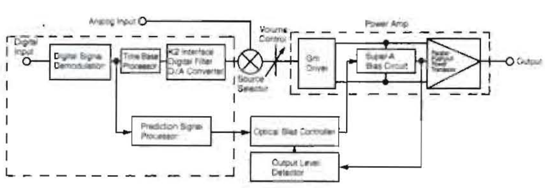


**Operation of Digital Pure-A II**

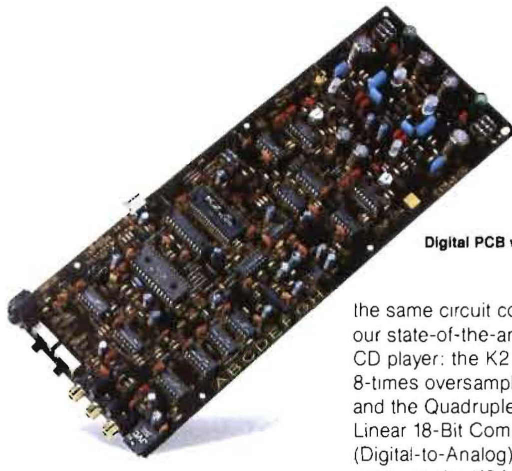
If there is no input, only minimum amounts of currents are applied (step 1). (In the figure above, the lower trace is the level of the upcoming signal, and the upper trace shows the level of bias current.) When the level, or dynamics, of the input signal increases to a moderate value—and this is the level you hear music at most of the time—the current increases to step 2, allowing you to enjoy the benefits of pure class-A operation. When the input signal level goes higher still, the current increases to the third, highest level at which you can enjoy both high power and pure class-A sound.

In Digital Pure-A II, rather than the power-supply voltage, the bias current fed to the power transistors is adjusted according to the level of the input signal, which translates into quicker response and higher efficiency. The bias current is always high enough to ensure that the amp operates in pure class-A, which reduces harmful harmonic distortion and provides pure sound. And the power-supply voltage for the power transistors is high enough to avoid clipping the waveform when a high-level signal is suddenly applied.

**Output power:** 100 watts per channel, min. RMS, both channels driven into 8 ohms, from 20Hz to 20kHz with no more than 0.004% total harmonic distortion

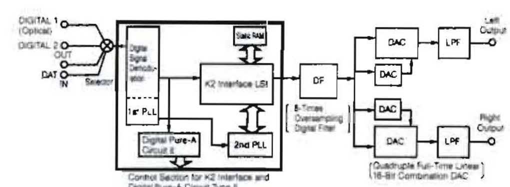


**Block Diagram of Digital Pure-A Circuit Type II**


**Digital PCB with K2 Interfac LSI**

*Built-in digital circuit: the K2 Interface, 8-times oversampling digital filter and quadruple D/A converters*

The AX-Z1010TN is a "digital" amplifier complete with a digital signal processing circuit for accepting and processing digital signals from your CD player, DAT (Digital Audio Tape) deck or other digital equipment. Here we've used the same circuit configuration a in our state-of-the-art XL-Z1010TN CD player: the K2 Interface, 8-times oversampling digital fil r, and the Quadruple Full-Time Linear 18-Bit Combination D/A (Digital-to-Analog) Converter. essence, the K2 Interface rem es ripple (waveform distortion) an jitter (shifts in time) by creating a totally new code. The oversamp ng digital filter removes re-quantizat on noise more effectively, while reducing group delay. And our exclusive combination D/A converter improves low-level linearity in order to reproduce delicacy and subtlety better. Mo details of these JVC technologie are found in the description of the XL-Z1010TN CD Player.



**Block diagram of AX-Z1010TN's digital circuitry**

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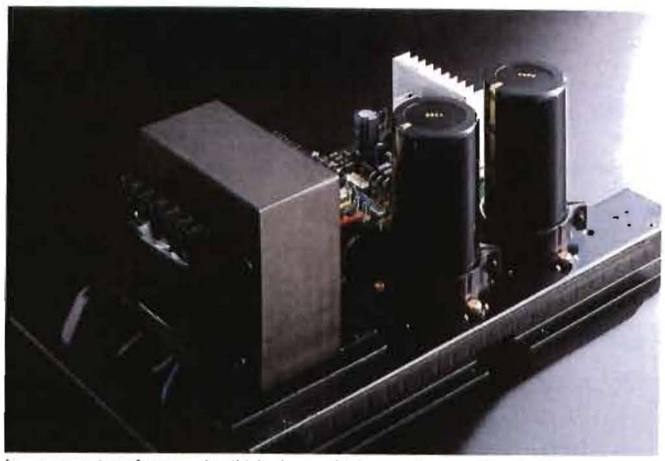


Large power transformer using thick wire and high-capacity capacitors for the power supply

### *Opt Super-A*

The AX-Z1010TN lets you enjoy class-A sound from analog programs, too, thanks to the Opt Super-A circuitry. There's a photocoupler device that controls the bias current to the power transistors in linear fashion, while preventing the transistors from switching off. As a result, you can enjoy smoother sound without crossover distortion, and at the same time the high efficiency of class-B amps.

### *Low-impedance driving capability*

Parallel Push-Pull Power Output

The dynamic sound of digital programs requires that power supply current be always available in abundance to cope with low impedances presented by high-power transients and complex loads. This ensures that digital programs are reproduced with low distortion and power to spare. The AX-Z1010TN is a "digital" amplifier for a reason: it's designed to capably handle load impedances as low as 4 ohms or even less. Indeed at 2 ohms it provides dynamic power as much as 320 watts per channel. We've made this possible in two ways. One, we've used high-performance, high-power transistors for the output devices and arranged them in parallel push-pull configuration. This reduces output impedance for improved speaker-driving capability and reduced distortion. Two, we've used thicker wiring in the power transformer to reduce power-supply impedance and improve the dynamic range.

### *Designs for pure signal transmission*


Optical and Coaxial Digital Inputs on Back

In the AX-Z1010TN, we've trimmed the signal path to the shortest possible length to maintain the high purity of input signals. We've achieved it using relays and electronic switches that let us avoid running connection wires from the back of the chassis to the front and then the back again. Moreover, there is a "DAC DIRECT" switch that when turned on allows the output from the D/A converter to go directly to the power amp after a very short trip, with only the volume control on line.

**Other features**

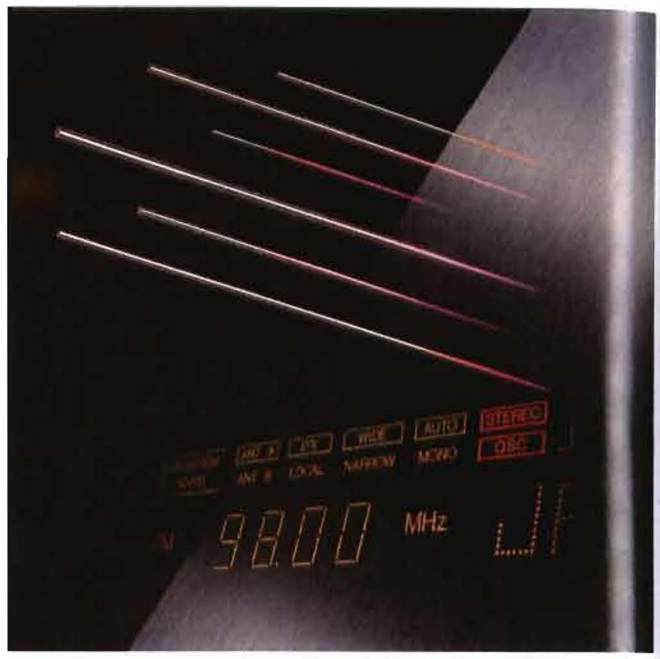
- •High-gain phono equalizer for MM/MC cartridges
- •Bass control
- •Remote control with volume control and speaker switching
- •Connections for 2 pairs of speakers

COMPU LINK Component

# FX-1010TN COMPUTER-CONTROLLED DIGITAL SYNTHESIZER TUNER

## *Computer control at its most accurate and convenient*

The primary function of a tuner is to bring in the station you want precisely and accurately, whether it's near or far. Using the latest circuit designs and devices, we've made our tuner from the SUPER DIGIFINE series, the FX-1010TN, more sensitive, selective and interference resistant than ever—the reason the tuner is able to provide wide dynamic range, low noise and low distortion, and wide frequency response. It's also extremely easy to use, thanks to the amazing power of an advanced microprocessor.

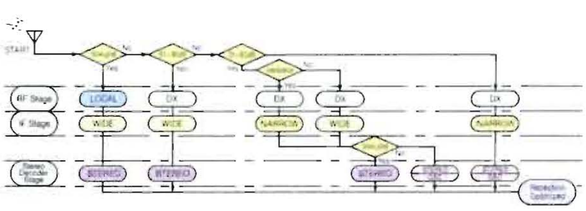



### *Reception servo for optimum reception*

The reception servo in the FX-1010TN ensures the best reception from any station, almost anywhere. A built-in micro-processor detects the strength of a tuned station and compares it against the degree of interference from adjacent stations. Then, depending on the degree of interference, the microprocessor selects the optimum operation mode for front-end, IF and multi-plex decoder stages (adjusting such parameters as RF gain, IF bandwidth, Quieting Slope Control and mono/stereo). Therefore, when interference is excessive, a narrow IF bandwidth is automatically chosen to prevent noise. And if there's no interference, then a wide IF bandwidth is automatically selected to give you remarkably clear sound. When a signal level is overly strong, the RF gain is reduced to avoid saturation distortion; when it's weak, sensitivity is increased.

Moreover, the tuner is equipped with inputs to connect two antennas; each may be oriented for the best reception of stations in diametrical opposition to the other. Conveniently, the position of antennas, A or B, may be stored in memory station by station, which allows most precise reception from any station, without multipath distortion.

To add to convenience, the selected parameters are clearly indicated on a large fluorescent display, letting you quickly confirm tuning status.



**Flowchart of Optimum Reception Servo System**

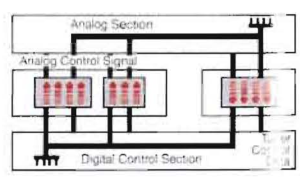
### *Features for low noise and low distortion*

JVC uses components and devices to ensure lowest possible noise and distortion and widest dynamic range from your favorite stations.

**"Opticalink" system**

The "Opticalink" system is one developed by JVC to ensure highest purity from your favorite stations. It uses eleven photocouplers (each consisting of a photoemitter and photosensor) to electrically decouple the analog from the digital section. As a result, interference between the two due to electric coupling is completely eliminated. This puts an end to digital noise and removes any trace of muddiness from the sound you hear.

In the FX-1010TN, each of the analog/tuner, digital control and "Opticalink" sections is mounted on its own PC board to shut out mutual interference and noise. And the digital control section, which can be a source of noise generation, is fully shielded to contain noise.

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Inside view of the FX-1010TN

### MOS FET front-end

The front end employs a dual-gate MOS FET which, along with high-Q varicaps, makes it as sensitive, selective and resistant to interference as a front end using an elaborate 5-ganged tuning capacitor. Therefore, the signal-to-noise ratio and interference-rejection capability are outstanding.

### Low-distortion IF section

The IF section of the FX-1010TN features ceramic filters coupled with a distortion reduction circuit for better phase response and low distortion.

### PLL detector

A PLL detector combines low distortion and high signal-to-noise ratio for dynamic and wide-range sound.

### Beatless PLL multiplex circuit

The FX-1010TN uses a new PLL multiplex circuit that electrically filters out beat noise—noise resulting from two or more frequencies interfering with one another. In a conventional CR-type MPX circuit using a VCO (Voltage-Controlled Oscillator), however, a device called an "anti-birdie filter" is used to remove beat noise, but the coils in the circuit tend to degrade signal quality. In our beatless multiplex circuit, moreover, a pilot canceller is featured to suppress the leakage of the pilot signal, thus obviating the need for a 19kHz filter, another component that would harm sound quality.

### Active filter

To prevent leakage of the 38kHz subcarrier contained in a stereo broadcast, an LC filter built from coils and capacitors is commonly used, but, due to the magnetic distortion the core of the coil causes, this system tends to degrade sound quality. In the FX-1010TN, however, an active filter has replaced the passive LC filter; with coils eliminated, it provides better high-frequency response, wider separation and higher signal-to-noise ratio.

## *Computer-controlled operating ease*

With the help of powerful microcomputers, we've also improved tuning ease of the FX-1010TN tremendously.


### Station name display

You can assign up to six alphanumeric characters to each preset station in the memory—"JAZZ-8," for instance.

### Auto memory

All 40 FM/AM stations can be automatically tuned in sequence and committed to memory as presets.

### Random preset memory

You can preset as many as 40 FM/AM stations in random order. There's a numeric keypad that allows direct access to any of 40 stations.

### Preset scan

All preset FM/AM stations can be automatically sampled one by one for approximately 5 seconds each.

### Preset cancel

Use this feature to skip undesired preset stations during preset scan.

### Program memory

Up to eight "events" (broadcasts) can be programmed for sequential recall under the control of an optional timer.

### Auto QSC

The Auto Quieting Slope Control circuit automatically goes into action to reduce noise when a station signal is weak.

### dB-referenced signal strength indicator

Read off the signal strength of a tuned station accurately down to 1dB—a convenience when orienting antennas.

### Variable stop level

Adjust FM/AM muting threshold in 5dB steps over a range during auto tuning. The variable stop level feature lets you adjust the threshold from 20dB to 60dB for FM, and from 60dB to 90dB for AM. When you use a higher level, you'll receive only powerful, clear-sounding stations, with weak stations muted out. Or, when you choose a low level, all receivable stations are tuned in.

### Record calibration signal generator

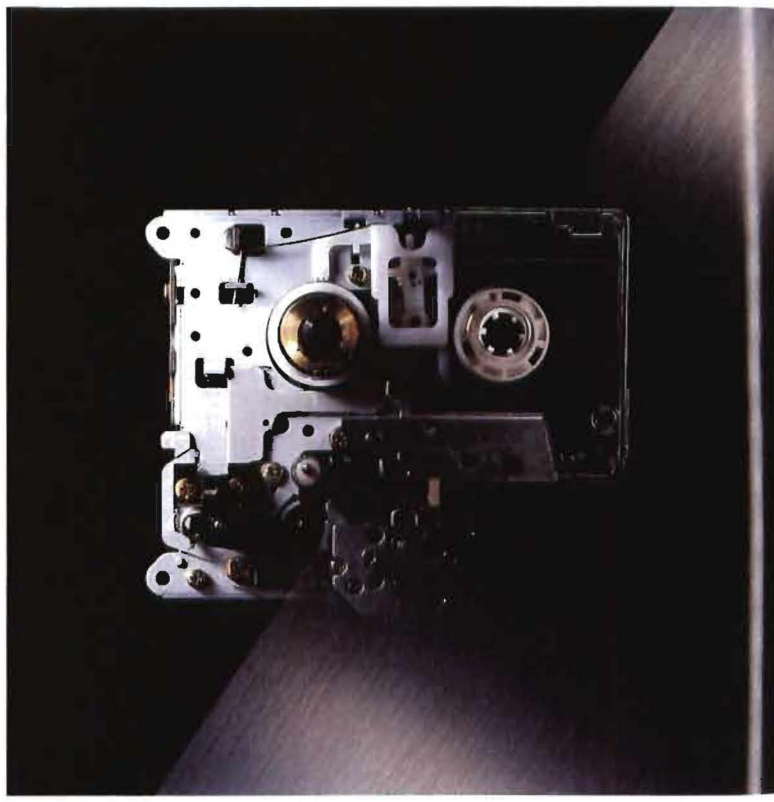
Record calibration signal generator outputs a standard 400Hz signal for recording level adjustment. So, you can easily set the recording level for different broadcasts or types of tape.

COMPU LINK
Component

# TD-V1010TN DISCRETE THREE-HEAD CASSETTE DECK

## *Getting closer to digital sound in performance*

It's true that audio equipment has been becoming more and more elaborate in circuitry these years. But the more complex circuitry is, the greater there are chances that the signals will be compromised in the process. So in the TD-V1010TN cassette deck from the SUPER DIGIFINE series, JVC took the classic "less is more" approach. Also in order to ensure pure and clean sound, JVC has employed a number of ways to damp and eliminate resonance and vibration.



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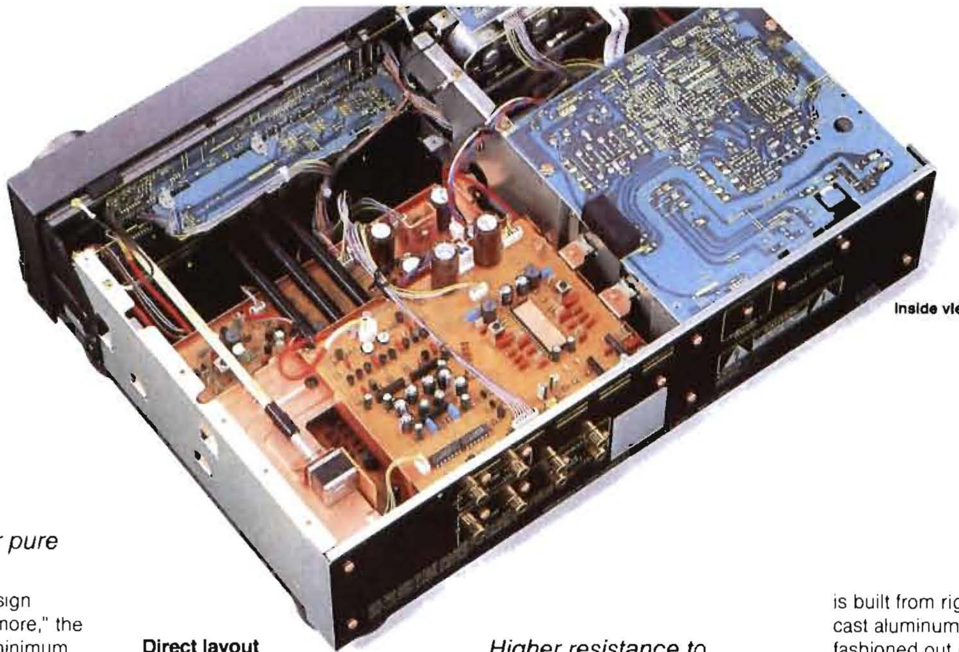


**Inside view of the TD-V1010TN**

## *"Direct" design for pure reproduction*

Based on the basic design philosophy of "less is more," the TD-V1010TN features minimum wiring and a signal path as direct and straightforward as technically possible.

**"DIRECT" inputs**

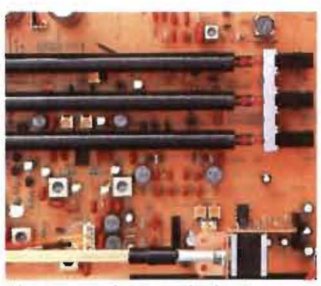

The TD-V1010TN has a "CD DIRECT" input—and a switch to select it—that lets you route the analog signal from a CD player direct to the cassette deck, entirely skipping circuitry in the receiver or amplifier. The TD-V1010TN also has a "DIRECT" input for connection of a second high-quality program source and the normal "LINE" input for connection with the amp or receiver, both with associated selector switches.

With "CD DIRECT" and "DIRECT" inputs, the applied signal runs the shortest route skipping the balance control. Even the Dolby HX-Pro and B/C noise reduction circuits can be completely put off line with the NR switches off. Trimmed wiring and fewer contacts add up to exceptional purity in the played back sound.

**Direct layout**

Also, the PC board for recording and playback is directly connected with input and output terminals, and the pattern is laid out symmetrically for the left and right channels to trim wiring. In addition, the input selector buttons, input level knob and power button on the front panel are linked via remote bars with switches and controls located near the input and output terminals on the rear. This of course is to cut the wiring to the bare minimum.

All this care to the last detail has resulted in reduction of noise, distortion and other sources of sound degradation that can compromise the purity of taped sound. This in turn means you can enjoy the fuller benefits of digital sound from our cassette deck—wide dynamic range and accurate recreation of most minute nuances, to name a few.



**Direct-connection separate circuits**
Input selector buttons and the input level knob are linked with switches and the control located at back via "remote bars."

## *Higher resistance to vibration and resonance*


Even the slightest vibrations and resonance can affect the quality of taped sound: when the tape is subjected to vibration during recording or playback, it can lead to modulation noise, the source of unclear, impure sound. So we've devised a number of ways to damp, suppress or totally eliminate them, ensuring pure reproduction.

**Vibration-suppressing mechanical designs**

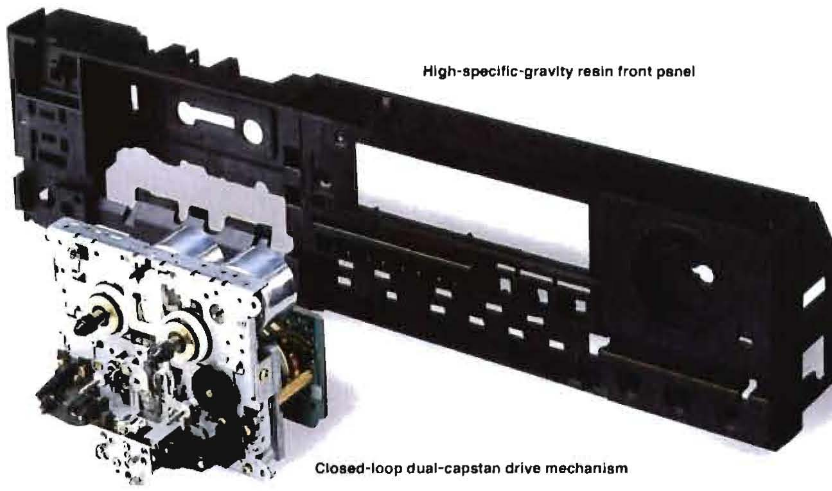
In the TD-V1010TN, the precision base for the tape drive mechanism is built from rigid and heavy die-cast aluminum. The front panel is fashioned out of non-resonating high-specific-gravity resin, which is 1.5 times heavier and 1.7 times more rigid than the conventional front-panel material. During recording and playback, the cassette is held firmly in place by a stabilizing pad that damps resonance and vibration. A metal sheet is attached to the circuit board supporting the motor to stop the flywheel from vibrating. Further, a massive brass weight is used for a reel receptacle to damp resonance and also improve rotating constancy. A solid and rigid base supports the entire chassis. Large insulators isolate the deck from sound pressure and external vibrations. And control knobs and switches are fashioned out of solid aluminum. This all adds up to pure and clean sound, especially from digital sound.

Combined record and play heads use "fine" amorphous material for superior high-frequency response and PCOCC wiring for signal purity



## Closed-loop dual-capstan drive with direct-drive motor

The TD-V1010TN employs perhaps the most sophisticated tape drive mechanism ever designed. With this drive, the portion of tape that runs across the heads is held taut during both recording and playback. One advantage of this sophisticated system is that it shuts out vibration and other external disturbances, resulting in reduced modulation noise. This gives you sound that is noticeably clearer and cleaner. And it improves head-to-tape contact for better response. The capstan is driven by a JVC precision high-torque coreless "pulse servo" motor which does not set up much vibration, further "cleaning" the taped sound.



High-specific-gravity resin front panel

Closed-loop dual-capstan drive mechanism

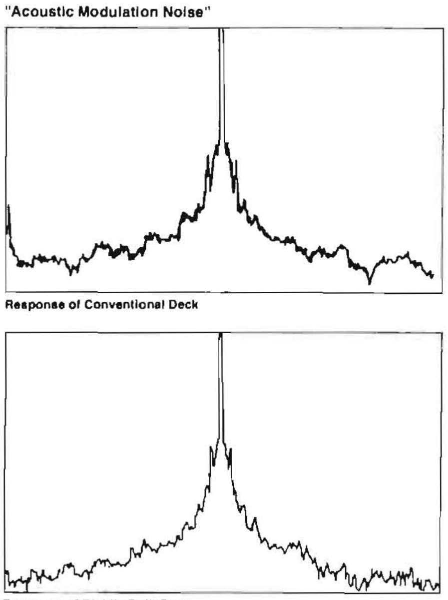
**Precision die-cast aluminum base for tape drive mechanism**

## "Acoustic modulation noise"

Up until now vibration and the degree of sonic degradation it can cause have not been correlated as meaningful data. So JVC has devised a new and more useful way to measure the effects of vibration on performance—the "acoustic modulation noise" test. In conventional modulation noise tests, the deck under test is set to simply record and play a 10kHz signal. But with our new method, the test deck is set to do so under 100-phon sound pressure, generated by a speaker reproducing pink noise. This setup simulates more closely the situation in which a deck is likely to be used at home.

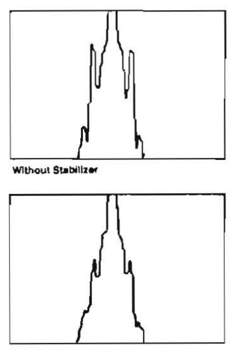
As you can see below, "acoustic modulation noise" is not only reduced overall in the TD-V1010TN, its curve is smooth. This proves that the TD-V1010TN has the capability to deliver exceptionally clean and transparent sound.

**"Acoustic Modulation Noise"**



Response of Conventional Deck

Response of Rigidly Built Deck



Without Stabilizer

With Stabilizer

**Improvement in Modulation Noise By Cassette Stabilizer**

## *3-head design for best response*

The TD-V1010TN features three independent heads, one each for recording, playback and erase. This means since the heads are separate, each can be given the optimum gap width for best response. Moreover, the azimuth, tilt and tracking can be aligned head by individual head for best results—extended response and low noise. This discrete 3-head construction also enables you to monitor recordings as they are being made. There are separate circuits for Dolby encoding and decoding, so you can monitor Dolby-encoded recordings in real time.

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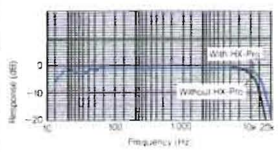
## *"Fine" amorphous heads using PCOCC wiring*

The record and play heads in the TD-V1010TN are made of newly developed "fine" amorphous material, which combines superior high-frequency response and our SA (Sen-Alloy) head's superior resistance to wear. For higher signal purity and clarity, we use PCOCC for the coils and leads in the heads. PCOCC is purest oxygen-free copper with single-crystal-like construction, featuring most efficient signal transmission and coloration-free sound reproduction.

## *Dolby* HX-Pro circuit*

Dolby HX-Pro is a new approach to improved high-frequency response. During recording, this circuit dynamically adjusts the bias according to the dynamic level of music, in order to improve the tape's saturation level and expand the dynamic range at high frequencies. In combination with Dolby B/C noise reduction systems, it allows a tape to record with a dynamic range as wide as that of digital programs. With most other decks, Dolby HX-Pro is undefeatable. But you can put it off line with the TD-V1010TN whenever you want higher purity.




**Frequency Response with and without Dolby HX-Pro**
The Dolby HX-Pro circuit improves the tape's MOL (Maximum Output Level) at high frequencies.

## *Elaborate circuitry for purer sound*

**DC-servo playback amp**
The playback amp features a low-noise FET differential input, cascade connection and DC servo system—an elaborate design usually featured in the phono equalizer of a quality amplifier. It has improved signal-to-noise ratio by 3 to 4dB.

**High bias frequency**
The bias frequency is twice higher than normal at 210kHz. This reduces beat noise due to interference of audio signals with the oscillator frequency.

**Low-impedance voltage-tracking power supply**
A highly regulated, low-impedance power supply is featured in the TD-V1010TN. Using a large-capacity transformer and electrolytic capacitors, it automatically tracks positive and negative voltages to keep the ground potential zero and improve the stability of amplifier circuits. Moreover, our power supply exhibits extremely low output impedance across the audible frequency range. The result? Even most dynamic musical passages or external noise cannot affect the steady performance of the power supply.

**Separate circuit construction**
In the TD-V1010TN, the audio amps, power supply and control circuits are mounted on separate PC boards, and insulated from each other, to shut out interference—both electric and magnetic. Furthermore, the deck uses circuit boards plated with OFC (Oxygen-Free Copper) to ensure that the most delicate nuances are reproduced precisely. Also contributing to cleaner sound is a fluorescent display off switch which turns off the display to prevent low-level digital noise.

**Other features**
- Computer-controlled full-logic control with "silent mechanism"
- Fluorescent Digital Peak Display, level meters and digital counter
- Auto rec mute, Music Scan, Timer start (record/play), Auto tape selector
- Bias and level controls for flat response and sensitivity matching
- Headphone output with volume control

*"Dolby" and the double-D symbol are trademarks of Dolby Laboratories Licensing Corporation.

# SX-911WD
# THREE-WAY SPEAKER SYSTEM

## *Another step closer to the "digital realism"*

For highly musical reproduction, especially of digital programs, high power alone is not enough. Therefore, in our speaker system for the SUPER DIGIFINE series, the SX-911WD, we've also improved linearity (especially at low levels), clarity, transparency, definition and depth. We have done so by upgrading our speaker-design technologies, from diaphragm unit to construction, to give you exciting "digital realism."

## Cloth-carbon woofer

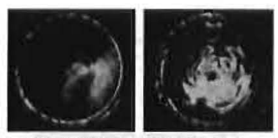
**Cloth-Carbon Diaphragm Material (Magnified)**

Our new cloth carbon woofer provides extended bass response, crispness and richness—properties thought to be mutually exclusive until now. This is because the newly developed cloth carbon, carbon fiber woven like cloth, features the ideal combination of light weight, high rigidity, high speed of sound and controlled internal loss.

Cloth carbon's high rigidity means that the vibrations of the voice coil are more exactly translated into the linear movement of the diaphragm, while the power range over which the diaphragm exhibits a piston-like motion (without cone breakup) is wider. Its controlled internal loss ensures an improved frequency response throughout the entire range the woofer covers. And the light weight means that the upper limit of the unit's frequency range is higher, extending the overall low-frequency response. Moreover, the surround of the woofer is made of a pliant material featuring quick response, which helps make the bass sound powerful, crisp and rich. The midrange also features "fine" cloth carbon for clear and natural sound.

## Amorphous-diamond coated tweeter

The amorphous-diamond coated tweeter provides exceptional transparency. This is because we use a totally new design for the tweeter: a dome diaphragm with a titanium base on which a thin layer of amorphous diamond is coated using a high-tech process called CVD (Chemical Vapor Deposition). Featuring uniform thickness, high purity and smoothness of surface, this coating increases the diaphragm's speed of sound to almost that of natural diamond. So the transient response is dramatically improved, as is sonic purity.



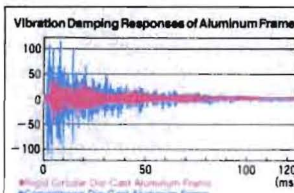
**Amorphous-Diamond Coated Diaphragm** **Titanium Diaphragm**

**Laser Holography Comparison**
Our hi-tech tweeter diaphragm is more resistant to cone breakup, retaining a piston-like motion into a much higher frequency range than do diaphragms made of conventional material

## Die-cast aluminium speaker frames

Every unit in the SX-911WD is housed inside a solid, unresonating die-cast aluminum frame. All frames are circular to disperse vibrations uniformly and efficiently. The woofer frame is especially impressive: it weighs 4.6 lbs. (2.1kg)—twice the weight of a common woofer frame. Firmly mounted by eight heavy-duty screws on the front baffle, the frame supports the moving structure so that rich and powerful bass of digital programs is played back with highest clarity. The solid frame also contributes to increasing the rigidity of the front baffle, protecting it from resonance and vibration. This ends spurious radiation, which means dramatic clarity in the reproduced sound.

**Vibration Damping Responses of Aluminum Frames**




**Rigid Pure-Aluminum Frame for SX-911WD Woofer**
A heavy die-cast aluminum frame mounts the speaker unit on the front baffle with eight solid screws. This configuration is extremely resistant to resonance.

**Cloth carbon woofer**

**"Fine" cloth carbon midrange**

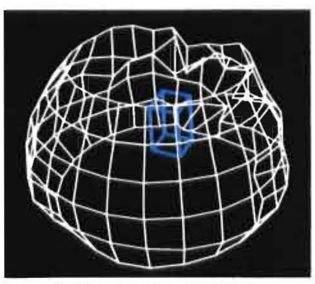
**Amorphous-diamond coated tweeter**

## *A solid enclosure with natural sonority*

The SX-911WD features a solidly constructed enclosure, using 1-inch (25mm) panel boards. To increase rigidity, front and rear baffles are mounted with additional cleats. And all sides of the enclosure are bonded together under one-ton pressure, making the entire enclosure as strong as it were made of a single piece of wood. This bonding process does not use heat, so the boards will stay firmly bonded for years, maintaining the enclosure's high rigidity. Moreover, the boards are made from high-density particleboard, chosen for its superb musical sonority.

## *Rounded front baffle*

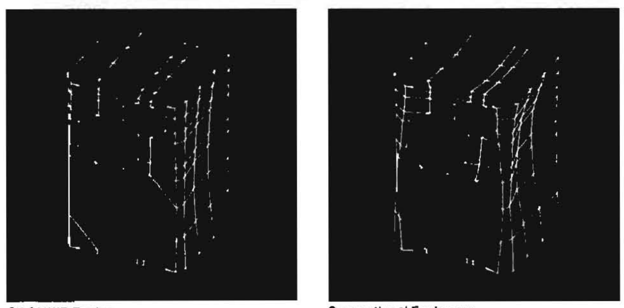
The round-cornered front baffle of the SX-911WD does more than lend class to overall design. It prevents the sound diffraction that can occur at sharp edges causing blurry and indistinct sound imagery. You'll enjoy clear definition and lifelike perspective from the SX-911WD.



**Propagation Characteristics of SX-911WD**

## *Computer simulations*

With JVC, the days of using trial-and-error methods to find the optimum positions for mounting speaker units on baffles are gone. Today, by inputting design parameters such as speaker unit characteristics, the physical properties of enclosure materials, and the sizes and shapes of front baffles, we can realistically simulate on computers how the sound will be generated and propagated. Thanks to this advanced technique, the SX-911WD combines better definition, smooth response and accurate phase response.



**SX-911WD Enclosure** **Conventional Enclosure**

**Vibration Analyses by Modal Technique**
Thanks to thick boards and rigid contruction, the enclosure of the SX-911WD is highly resistant to even the minutest deformation and resonance.

## *Quality parts*

We use only quality parts in the SX-911WD so that the sound, whether digital or analog, is the best you have ever heard. For instance, the frequency-dividing network is made from quality parts to minimize signal loss and phase distortion. Moreover, the network is also physically divided into three parts—one each for the high, mid and low frequency range—to shut out interference. Furthermore, wires are not soldered but clamped firm and then hermetically sealed by a bonding agent. This ensures minimum degradation of the signal over longer periods of time.

- High power handling capacity: 150 watts/300 watts (Music)
- 12-inch (30.5cm) cloth carbon woofer for the bass sound that's crisp, extended and rich
- 4-1/2-inch (11.5cm) "fine" cloth carbon midrange for rich and natural midrange sound
- 1-inch (2.5cm) amorphous-diamond coated tweeter for transparency and superior transient response
- Frequency response: 40—50,000Hz
- Sensitivity (1m on axis): 91dB/W

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# SX-911WD
THREE-WAY SPEAKER SYSTEM
SUPER DIGIFINE

**COMPU LINK** A component that incorporates the JVC COMPU LINK Control System. This system permits it to "talk," or operate interactively, with other COMPU LINK Components, allowing one-touch source selection/playback, or coordinated operation.

**COMPU LINK** A component that has versatile remote control functions built-in. Its COMPU LINK Remote Control System allows individual COMPU LINK Components to be operated from a single remote control unit.

# SPECIFICATIONS

## XL-Z1010TN
### Compact Disc Player

| | |
|---|---|
| Frequency Response | 2Hz — 20kHz |
| Total Harmonic Distortion (1kHz) | 0.003% |
| Dynamic Range (1kHz) | 99dB |
| Signal-to-Noise Ratio | 108dB |
| Channel Separation (1kHz) | 102dB |
| Wow and Flutter | Unmeasurable |
| Output Level | 2.0V RMS |
| Dimensions (W×H×D) | 18-3/4×4-9/16×14-13/16 inches<br>475×115×375mm |
| Weight | 16.4 lbs. (7.4kg) |

## RX-1010VTN
### Receiver

| | |
|---|---|
| **AMPLIFIER SECTION** | |
| Output Power:<br>2-Channel Operation | **120 watts per channel, min. RMS, both channels driven into 8 ohms from 20Hz to 20kHz, with no more than 0.007% total harmonic distortion** |
| 4-Channel Operation | (Front Channels)<br>**110 watts per channel, min. RMS, both channels driven into 8 ohms from 20Hz to 20kHz, with no more than 0.007% total hermonic distortion**<br>(Rear Channels)<br>15 watts per channel, min. RMS, into 8 ohms at 1kHz, with no more than 0.07%* total harmonic distortion |
| Dynamic Power**<br>2 ohms/4 ohms/8 ohms | 360 watts/260 watts/190 watts |
| Total Harmonic Distortion<br>(8 ohms, 1kHz) | 0.003%* at 125 watt output |
| Intermodulation Distortion | 0.007% at 120 watt output |
| Input Sensitivity/Impedance<br>PHONO MM<br>PHONO MC<br>VIDEO SOUND/AUX/CD/TAPE | <br>2.5mV/47k ohms<br>250μV/100 ohms<br>230mV/47k ohms |
| Signal-to-Noise Ratio ('66 IHF/78' IHF)<br>PHONO<br>VIDEO SOUND/AUX/CD/TAPE | <br>80dB/80dB (REC OUT)<br>100dB/85dB |
| Frequency Response<br>PHONO<br>VIDEO SOUND/AUX/CD/TAPE | <br>20Hz — 20kHz (±0.5dB)<br>5Hz — 50kHz (+0, −1dB) |
| RIAA Phono Equalization | ±0.5dB (20Hz — 20kHz) |
| Loudness Control<br>(Volume control at −30dB position) | +6dB at 100Hz, +4dB at 10kHz |
| **S.E.A. SECTION** | |
| Center Frequencies | 63, 160, 400, 1k, 2.5k, 6.3k, 16kHz |
| Control Range | ±10dB |

| | |
|---|---|
| **FM TUNER SECTION** (IHF) | |
| Usable Sensitivity | 10.8dBf (0.95μV/75 ohms) |
| 50dB Quieting Sensitivity: MONO<br>STEREO | 16.3dBf (1.8μV/75 ohms)<br>38.3dBf (22.5μV/75 ohms) |
| Stereo Separation at REC OUT (1kHz) | 40dB |
| Distortion (1kHz) MONO/STEREO | 0.15%/0.2% |
| Signal-to-Noise Ratio (IHF-A Weighted)<br>MONO/STEREO (at 85dBf) | 81dB/73dB |
| Selectivity (±400kHz) | 65dB |
| Capture Ratio (10mV/300 ohms) | 1.5dB |
| IF Response Ratio (98MHz) | 85dB |
| Frequency Response | 30Hz — 15kHz (+0.5, −3dB) |
| **AM TUNER SECTION** | |
| Usable Sensitivity | 300μV/m (Loop antenna)<br>30μV (External antenna) |
| Signal-to-Noise Ratio (100mV/m) | 50dB |
| Selectivity (±10kHz) | 38dB |
| **VIDEO INPUTS/OUTPUTS** | |
| Output Signal Level | 1Vp-p (at 1Vp-p input) |
| Impedance | 75 ohms unbalanced |
| Synchronization | Negative |
| Signal-to-Noise Ratio | 45dB |
| Crosstalk | 45dB (3.58MHz) |
| Dimensions (W×H×D) | 17-3/16×6-3/16×15-1/2 inches<br>435×156×393mm |
| Weight | 27.6 lbs. (12.5kg) |

*Measured by JVC Audio Analysis System.
**EIA Dynamic Test Signal.

## XP-A1010TN
### Digital Acoustics Processor

| | |
|---|---|
| Volume Control | 6-ganged motor-driven remote-controlled volume control |
| D.A.P. Level Volume Control | Front/Rear |
| A/D Converter | 16-bit linear, 48kHz sampling, 64X oversampling digital filter |
| D/A Converter | 16-bit linear, 4X oversampling digital filter |
| Number of Sound Field Patterns | Programmed: 20<br>Manual: 20 |
| Input Terminals: ANALOG<br>DIGITAL OPTICAL<br>DIGITAL COAXIAL | LINE IN/TAPE PLAY<br>OPTICAL LINE IN<br>COAXIAL LINE IN/COAXIAL DAT PLAY (32k/44.1k/48kHz)<br>3-mode automatic selection |
| Output Terminals: ANALOG<br>DIGITAL OPTICAL<br>DIGITAL COAXIAL | MAIN/FRONT/REAR<br>OPTICAL LINE OUT<br>COAXIAL LINE OUT/COAXIAL DAT REC |
| Tape Functions | NORMAL/PLAYBACK/D.A.P. REC |
| Level/Impedance: IN<br>OUT | 500mV/47k ohms (LINE IN, TAPE PLAY)<br>5V/660 ohms (F/D.A.P. OUT, R/D.A.P. OUT) |
| Total Harmonic Distortion: MAIN OUT<br>D.A.P. OUT | 0.002% (1kHz, 2V output)<br>DIGITAL: 0.004%/ANALOG: 0.005% (1kHz, 2V output) |
| Frequency Response: MAIN OUT<br>D.A.P. OUT | 5Hz — 100kHz (+0, −3dB)<br>DIGITAL: 5Hz — 20kHz (+0.5, −1dB)<br>ANALOG: 5Hz — 20kHz (+0.5, −3dB) |
| Dynamic Range: D.A.P. OUT | DIGITAL: 94dB/ANALOG: 94dB |
| Signal-to-Noise Ratio: MAIN OUT<br>D.A.P. OUT | 110dB<br>DIGITAL: 100dB/ANALOG: 94dB |
| Dimensions (W×H×D) | 17-3/16×4×14-3/16 inches<br>435×101×360mm |
| Weight | 12.8 lbs. (5.8kg) |

# AX-Z1010TN
## Amplifier

| | |
|---|---|
| **CIRCUITRY** | |
| Phono EQ Amp | ICL MM/MC equalizer with EL-FETs in its initial stage |
| Power Amp | Digital Pure-A II/Opt Super-A power amplifier with Gm Circuit |
| Power Supply | "Clean & Dynamic" power system for power amplifying stage |
| **OVERALL CHARACTERISTICS** | |
| Output Power | **100 watts per channel, min. RMS, both channels driven into 8 ohms from 20Hz to 20kHz, with no more than 0.004% total harmonic distortion**<br>105 watts per channel, min. RMS, into 8 ohms at 1kHz, with no more than 0.002%* total harmonic distortion |
| Dynamic Power**<br>2 ohms/4 ohms/8 ohms | 320 watts/230 watts/135 watts |
| Total Harmonic Distortion<br>AUX to SP OUT<br><br>PHONO to SP OUT | <br>0.004% at 100 watt output, 8 ohms, 20Hz to 20kHz<br>0.002%* at 105 watt output, 8 ohms, 1kHz<br>0.009% at 100 watt output, 8 ohms, 20Hz to 20kHz, −20dB volume |
| Intermodulation Distortion<br>(60Hz:7kHz=4:1) | 0.004% at 100 watt output, 8 ohms |
| Switching Distortion | 0 |
| Transient Intermodulation Distortion | 0 (LPF fc=100kHz) |
| Power Bandwidth | 5Hz to 60kHz (IHF, both channels driven, 8 ohms, 0.03% total harmonic distortion) |
| Frequency Response (8 ohms)<br>TUNER/AUX/CD/TAPE | 5Hz to 100kHz +0dB, −3dB |
| Damping Factor (1kHz, 8 ohms) | 200 |
| REC Output Level/Impedance | 300mV/1k ohms<br>2.0V/1k ohms (DIGITAL) |
| Input Sensitivity/Impedance (1kHz)<br>PHONO MM<br>PHONO MC<br>TUNER/AUX/CD/TAPE | <br>4mV/47k ohms<br>300µV/470 ohms<br>300mV/30k ohms |
| Signal-to-Noise Ratio ('66 IHF/'78 IHF)<br>PHONO MM<br>PHONO MC<br>TUNER/AUX/CD/TAPE | <br>89dB/82dB (REC OUT)<br>71dB (250µV input)/73dB (REC OUT)<br>112dB/86dB |
| Loudness (−30dB Volume) | +5dB at 50Hz |
| **PHONO EQUALIZER SECTION** | |
| Phono Overload (1kHz) MM<br>MC | 100mV (0.02% total harmonic distortion)<br>7mV (0.03% total harmonic distortion) |
| RIAA Phono Equalization MM<br>MC | ±0.2dB (20Hz to 20kHz)<br>±0.2dB (20Hz to 20kHz) |
| **D/A CONVERTER SECTION** | |
| Sampling Frequencies (Auto Selection) | 32k, 44.1k, 48kHz |
| Total Harmonic Distortion (1kHz) | 0.0035% |
| Dynamic Range (1kHz) | 98dB |
| Signal-to-Noise Ratio | 107dB |
| Digital Input/Output Terminals<br>OPTICAL<br>COAXIAL | <br>−23 – −14dBm<br>0.5Vp-p/75 ohms |
| Dimensions (W×H×D) | 17-3/16×6-13/16×18-1/8 inches<br>435×173×459mm |
| Weight | 37.1 lbs (16.8 kg) |

*Measured by JVC Audio Analysis System.
**EIA Dynamic Test Signal.

# FX-1010TN
## Tuner

| | |
|---|---|
| **FM TUNER SECTION** (IHF) | |
| Usable Sensitivity | 10.3dBf (0.9µV/75 ohms) |
| 50dB Quieting Sensitivity<br>MONO<br>STEREO | <br>14.8dBf (1.5µV/75 ohms)<br>38.1dBf (22µV/75 ohms) |
| Signal-to-Noise Ratio (IHF-A Weighted)<br>MONO/STEREO (at 85dBf) | 94dB/88dB |
| Total Harmonic Distortion (1kHz)<br>MONO/STEREO | 0.009%/0.02% (WIDE) |
| Capture Ratio | 1.2dB |
| Selectivity (±400kHz) | 25dB (WIDE)/75dB (NARROW) |
| IF Response Ratio | 110dB |
| AM Suppression Ratio | 65dB |
| Stereo Separation (1kHz) | 60dB (WIDE) |
| Frequency Response | 20Hz – 15kHz (+0.3, −0.5dB) |
| Antenna Input Impedance | 75 ohms unbalanced×2 |
| Output Signal Level | 600mV (2.2k ohms) |
| REC CAL Output Level | Equivalent to 50% FM modulation |
| **AM TUNER SECTION** | |
| Usable Sensitivity | 250µV/m (Loop antenna) |
| Total Harmonic Distortion | 0.3% |
| Signal-to-Noise Ratio | 50dB |
| Selectivity (±10kHz) | 35dB |
| Image Response Ratio | 40dB |
| IF Response Ratio | 60dB |
| Dimensions (W×H×D) | 17-3/16×3-15/16×11-3/4 inches<br>435×100×298mm |
| Weight | 8.2 lbs (3.7kg) |

# TD-V1010TN
## Cassette Deck

| | |
|---|---|
| Frequency Response<br>At −20dB<br>Metal Tape<br><br>SA Chrome Tape<br><br>Normal Tape | <br><br>10 – 22,000Hz<br>(15 – 20,000Hz ±3dB)<br>10 – 20,000Hz<br>(15 – 18,000Hz ±3dB)<br>10 – 20,000Hz<br>(15 – 18,000Hz ±3dB) |
| Signal-to-Noise Ratio | 61dB●(Metal) |
| Wow and Flutter | 0.022% (WRMS) |
| Crosstalk (1kHz) | 65dB |
| Channel Separation (1kHz) | 40dB |
| Harmonic Distortion<br>Total (0VU, 1kHz)<br>K3 (0VU, 1kHz) | <br>1.0% (Metal)<br>0.5% (Metal) |
| Heads: Record/Playback<br>Erase | Discrete 3-head configuration, fine amorphous heads for recording and playback<br>Two-gap ferrite head |
| Motors | Pulse Servo direct-drive motor<br>DC motor×2 |
| Input Sensitivity/Impedance<br>Line Input | 80mV/50k ohms (Direct input×2) |
| Output Level/Impedance<br>Line Output<br>Headphones | <br>300mV/600 ohms<br>0 – 1mW/8 ohms (Matching impedance: 8 – 1k ohms) |
| Dimensions (W×H×D) | 17-3/16×5-9/16×13-1/4 inches<br>435×140×336mm |
| Weight | 22.7 lbs (10.3kg) |

● Measured at peak level, weighted, without NR. The S/N is improved by about 15dB at 500Hz and by about 20dB above 1kHz with Dolby-C NR on, and by 5dB at 1kHz and by 10dB above 5kHz with ANRS/Dolby-B NR on.

# SX-911WD
## Speaker System

| | |
|---|---|
| Type | 3-way, acoustic suspension |
| Speakers<br>Woofer<br>Midrange<br>Tweeter | <br>12" (30.5cm), cloth carbon cone<br>4-1/2" (11.5cm), cloth carbon cone<br>1" (2.5cm), amorphous-diamond coated dome |
| Power Handling Capacity | 150 watts<br>300 watts (Music) |
| Impedance | 6 ohms |
| Sensitivity (1m on axis) | 91dB/W |
| Frequency Range | 40 – 50,000Hz |
| Crossover Frequencies | 500Hz, 4kHz |
| Dimensions (W×H×D) | 15×26-3/16×13-7/8 inches<br>380×665×351mm |
| Weight | 62.8 lbs (28.5kg) |

JVC®

**JVC COMPANY OF AMERICA**
DIVISION OF US JVC CORP.
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